nereus

User Manual
Ver 5.4.1.2

PROCESAMIENTO DIGITAL Y SISTEMAS, S.L

Index

CE Declaration of Compliance

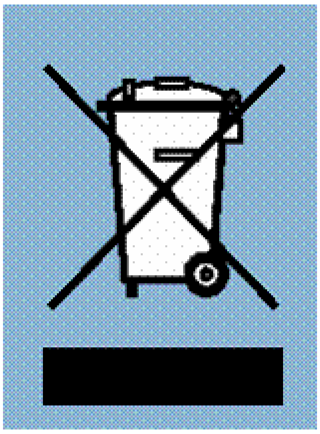
Procesamiento Digital y Sistemas S.L., hereby declares that **Nereus** bearing the CE168X parking are in compliance with Electromagnetic Compatibility Directive (89/336/EEC), and the Low Voltage Directive (72/23/EEC) of the European Union.

A "Declaration of conformity" for **Nereus** is available on file at Prodys offices in Spain. To obtain this information, contact with sales@prodys.net.

CAUTION

Nereus uses a Lithium battery.

Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the manufacturer. Dispose of used batteries according to the manufacturers instructions.



Your product is designed and manufactured with high quality materials and components, which can be recycled and reused.

When this crossed-out wheeled bin symbol with black bar underneath is attached to a product it means that product is covered by the European Directive 2002/96/EC.

Please, inform yourself about the local separate collection system for electrical and electronic products.

Please act according to your local rules and do not dispose of your old products with your normal household waste. The correct disposal of your old product will help prevent potential negative consequences for the environment and human health.

Chapter I

About this manual

The information is arranged in the following sections:

- **Chapter II – Overview.**
  This chapter describes the main features of Nereus and its components.

- **Chapter III – Installation Guide.**
  This chapter provides hardware requirements and instructions for installing the **Nereus** unit.

- **Chapter IV – Nereus Web Browser.**
  To set up and configure the system, you need to connect **Nereus** via Ethernet to a PC running the web browser. This chapter describes how to start it and how to use it.

- **Appendix A – Technical Specifications.**

- **Appendix B – Updating the firmware.**
  This appendix describes how to update the Nereus firmware.

Chapter II

Overview

Nereus has been designed to provide a high density Audio over IP platform. The system is based on a 19” 3U chassis which can house up to 14 IP audio codec cards or up to 7 couples of ISDN/X21+IP cards. Each IP audio codec card is based on the successful ProntoNet IP codec, which mechanic has been adapted to be connected as a compactPCI card. All features present in a ProntoNet are available on Nereus, providing the best space-saving solution when many IP audio codecs are required.

All modules support **hot-swapping** allowing live insertion and extraction while the unit is running. This means that it is possible dynamic reconfiguration and easy maintenance.

Nereus comprises power supplies modules and the numbers of audio cards that users need. Two power supplies modules can be fitted, one as Main and one as Slave, when redundant power supply is required. The switch over between both is totally transparent to any active communications at that moment.

Nereus supports individual remote control of each IP audio codec via embedded web server or global system centralized management via ProdysControl application including a centralized alarms receiving centre.

II.1 Key Features

- **General**
  - Configurable and scalable system.
  - Up to 14 independent Audio over IP codecs (**ProntoNet** in compactPCI format).
  - Up to 7 couples of IP+ISDN/X21 cards (7 independent ISDN/X21/IP audio codecs).
  - Modular design: Hot-swappable, plug-in modules allowing for easy upgrades and system expansion.
  - Redundant power supply.
  - Centralized management from Prodys Control application.
  - High quality audio transport over IP networks.
  - Like ProntoNet, each card of **Nereus** uses a special real-time operation system running on dedicated microprocessor. There is no PC-based system inside the **Nereus**.

- **Audio Interfaces**
  - Analog (24 bits A/D and D/A converters) and digital AES/EBU interfaces.
  - Dolby E over IP (transparent mode).
  - AES/EBU User bit transmission.

- **Audio Compression**
  - Widest range of industry-standard coding algorithms:
    - G711 A/µ Law.
    - G722.
    - MPEG 1,2 LII (ISO/IEC 11172-3 /13818-3).
    - MPEG 1,2 LIII (ISO/IEC 11172-3 /13818-3).
    - MPEG 2 AAC LC (ISO/IEC 13818-7).
    - MPEG 4 AAC LC (ISO/IEC 14496-3).
    - MPEG 4 AAC LD (ISO/IEC 14496-3).
    - MPEG 4 AAC HE (ISO/IEC 14496-3).
    - Standard and Enhanced aptX™.
    - PCM (uncompressed linear audio).

- **Communications**

  - 10/100 BT Ethernet port for audio streaming and control.
  - ISDN/X21 interface card as option.
  - Unicast, Multicast and Multi-Unicast audio streaming transmission/reception.
  - IP compatibility according to the standard defined by the N/ACIP project within the EBU group.
  - IP protocols and standards supported:

- RTP: Real Time audio streaming.
- SIP: Session establishment.
- SAP: Session announcement protocol for multicast communications.
- SDP: Session description.
- MIME: Standard payload definitions.
- DHCP: IP address auto configuration.
- SNMP: Remote monitoring over SNMP protocol.
- HTTP: Embedded Web server. Full real time control and monitoring
- RIP2: Internal/Alias IP addresses can be assigned.
- IGMPv2: Multicast groups membership.
- SMTP: Alarm monitoring via e-mail.
- SNTP: Clock synchronization.
- DNS: Domain Name Server.

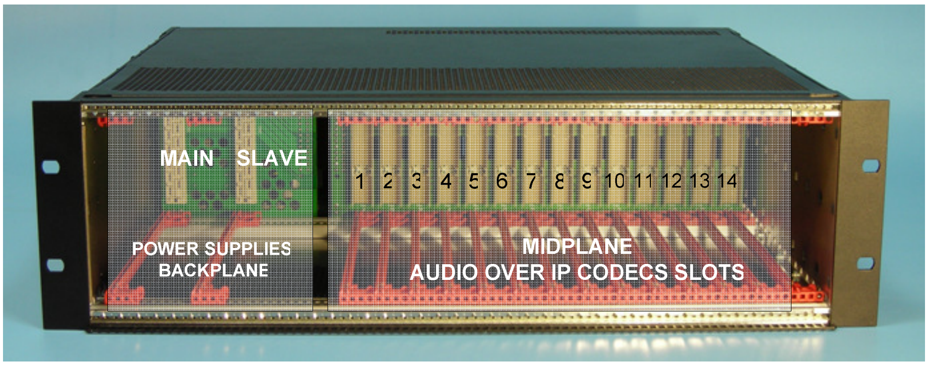
- **Streaming advanced features**
  - Jitter correction buffer.
  - Automatic delay compensation.
  - IP packet size configurable.
  - Test tool to check bandwidth, delay, jitter and QoS parameters.
  - Real Time Analyzer to check jitter, buffer usage and quality of service concurrently to the audio connection.

- **Auxiliary Data**
  - RS232 port for ancillary data.
  - GPIO port: 4 inputs (contact closures to ground) and 4 open collector outputs.

- **Remote Control and Monitoring**
  - Centralized management via embedded web server.
  - SNMP and e-mail alarm notification.
  - Customizable remote control and monitoring via Proprietary Control Protocol over RS232 or IP.

II.2 Nereus Architecture

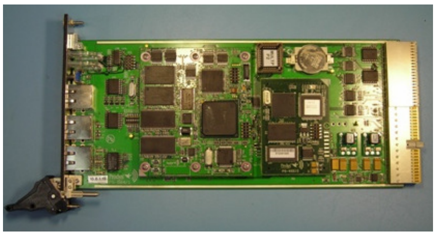
Nereus architecture is based on a midplane where can be accommodated up to 14 audio codec cards and a backplane for one or two power supplies. Both midplane and backplane are passive to help ensure high reliability, that is, they contain no active logic, just connectors and traces. All cards are hot-swappable and can be installed without interrupting the operation of the system.

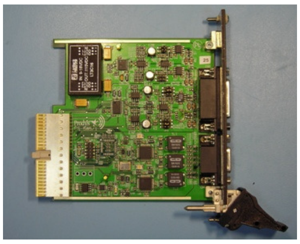


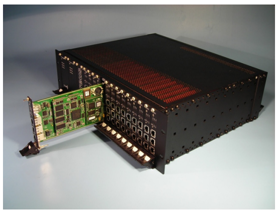
The midplane performs the following functions:

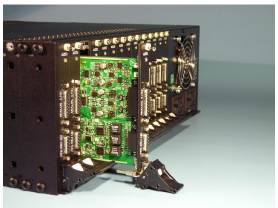
- Provides a mechanical connection for all codecs cards.
- Provides the 12 VDC power from the power supply to each codec card.
- Provides the interconnection between the codec card and the audio interfaces located on the rear panel.

Consequently, each audio over IP codec card actually consists of two cards: the front card which is in charge of the communications and the transition card with the audio interfaces:



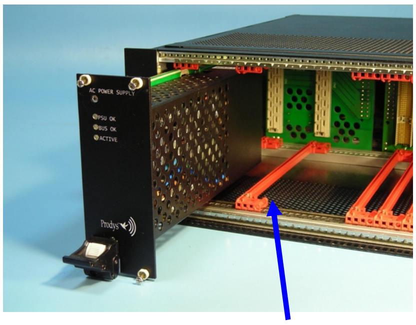






Each audio over IP codec card is totally independent from each other and there is not a control module to manage all of them. Therefore, it is necessary to connect the Ethernet interface of any card in order to gain access to it. It is possible to install one ISDN/X21 card per IP Doris card (up to 7 couples). This ISDN/X21 card does not require the installation of any additional audio I/O card.

Nereus can accommodate up to 2 power supplies. The base system comes fitted with one 200 watt power supply as standard, but optionally one power supply can be added to provide redundancy in the event of a power supply failure.
The power supply requires no configuration but it is possible to monitor their status through the web page of each card or from the Prodys Control application.

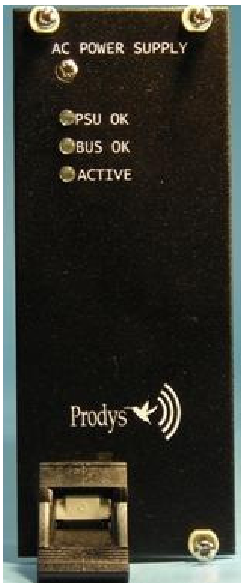


Slot for redundant power supply

II.3 Nereus Components

II.3.1 Main power supply.

The same power supply can work as Master or Slave and it is the slot where is installed which decides the operation mode: if the power supply is installed in the left most slot, it will work as Main power supply and if it is installed in the second one, it will work as secondary power supply.
Power supply requires no special setup. As long as it is installed (plugged) into the slot and power is applied, it is operating. It is possible to monitor its status from the web browser control of each card or from the Prodys Control application.



LED's description:

PSU OK	AC power present	• Green → AC power is available to the Power supply. • Orange → AC power supply is NOT available to the Power supply.
BUS OK	Backplane power supply	• Green → Backplane power supply OK. • Off → Power supply backplane fail.
ACTIVE	Power supply active	• Green → Power Supply operating. • Orange → Power supply as backup.

On the rear panel there are two separated AC power inlets, one for the Main power supply and the other one for the secondary or backup power supply.

II.3.2 Secondary power supply:

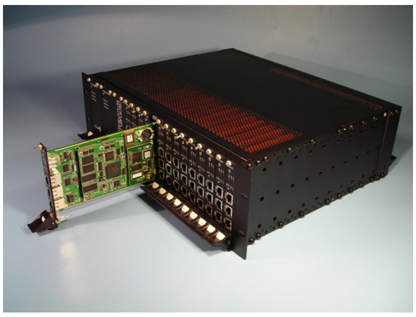
Each Nereus system can contain a second power supply for redundancy. If the main supply fails, the system will continue operating using the redundant power supply. To install a redundant power supply, simply insert the second power supply into the redundant power supply slot (its LED indicators will light up identically to those in the main supply).

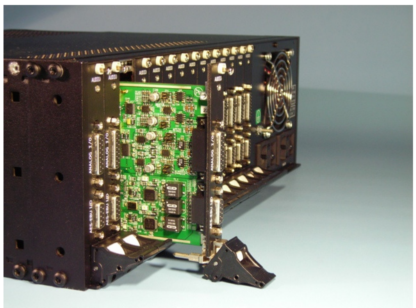
When a power supply fails it may be removed and a new power supply module inserted without powering down the system. Given that there are independent power connectors for each power supply, the insertion of a new power supply with its plugging connected must be avoided.

It is always installed in the second slot of the power supply backplane.

II.3.3 Audio over IP codec cards:

The audio over IP codec cards are totally independent and contain all the necessary interfaces to work with them. Each codec actually consists of two cards, one is installed on the front panel and the other one with the audio connections, on the rear panel. There are 14 slots in the midplane where they can be installed.

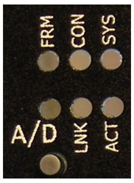




The front panel connectors and LED's are the following:

- LAN connector: The LAN socket is a standard 100Base-Tx (10/100 Mbps) Ethernet connection that takes a RJ45 plug. Through this Ethernet port it is possible to transmit and receive audio, as well as manage the equipment.
- RS232 connector: It allows the transmission and reception of auxiliary data along with the encoded audio in IP communications. More information on chapter IV.7.11 – Aux. Data.
- GPIO connector: The GPIO port connector allows remote control/signalling by means of remote contact closures. There are four ground contact inputs and open collector outputs. More information on chapter III.5.3 – GPIO port.

LED's Description:



	Functionality	
FRM	Decoder status	• Green → Decoder framed. • Off → Decoder NOT framed.
CON	Communication status	• Green → Line connected. • Off →
SYS	System status	• Green → Normal operation. • Green blinking → Booting process. • Red → Alarm activated. • Red blinking → Sw updating in process. • Orange → Past alarm.
A/D	Audio input selected	• Green → Analog input selected. • Red → Digital input selected.
LNK	LAN connection status	• Green → LAN connected (physical level detected). Good connection between the card and network. • Red → LAN disconnected. No connection between card and network.
ACT	Rx LAN activity	• Green → Data from the LAN detected. • Off → No data detected.

There is a mechanical switch close to the LED's to select manually between analog or digital audio input.

On the rear panel the audio interface cards are installed, with two separated connectors for analog and digital interfaces.

II.3.4 Audio over ISDN/X21 cards:

The audio over ISDN/X21 card is optional and contains the necessary interfaces to stream audio over ISDN and leased lines (X21). Each ISDN/X21 card must be plugged along with an IP card. The IP card must be on the left side of the ISDN card. Therefore, up to 7 couples of IP+ISDN/X21 cards can be hosted by Nereus.

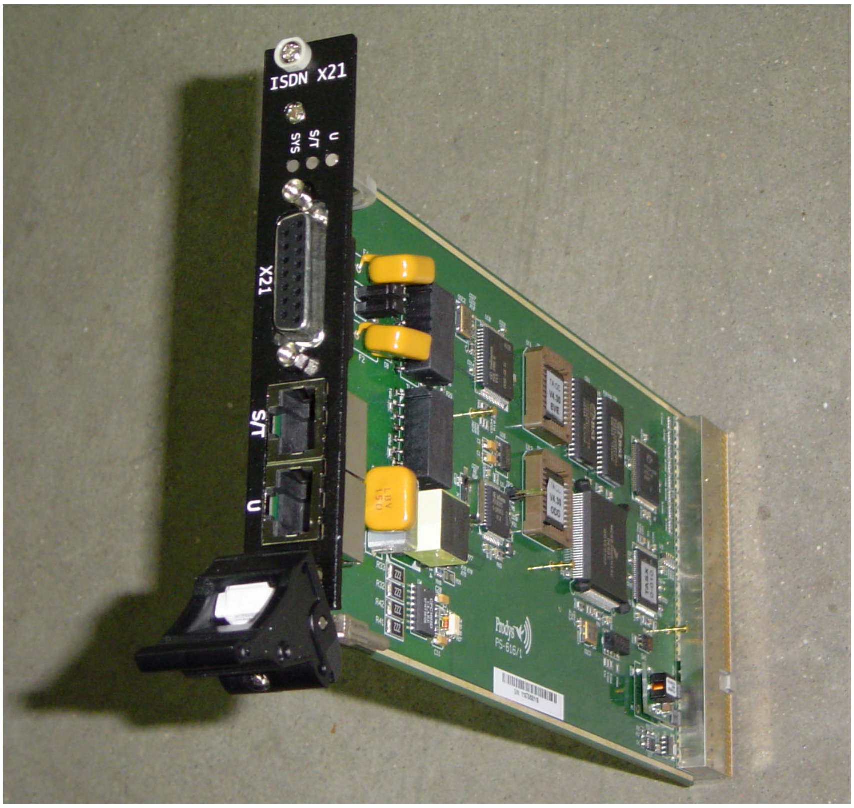
The front panel connectors are the following:

ISDN

- Protocols: EISDN, AT5ESS, DMS100 and NAT.
- 1 BRI connection. S/T and U interfaces.
- **BackUp system:** ISDN as a BackUp for IP or X21 links.
- RJ45 connector.

X21 Port

- Serial Synchronous interface.
- Bit rates: 64, 128, 192, 256, 384 and 576kbps.
- 15 Ways DB connector.



Chapter III

Installation Guide

This chapter describes Nereus hardware and user installation.

The installation and servicing instructions in this manual are for use by qualified personal.

III.1 Initial checks

Before unpacking unit check its packaging for any signs of damage or mishandling during transportation, report any damage to the shipping company immediately. Unpack the unit carefully, if you find any damage or the unit does not work correctly, you should contact Prodys or its distributor as soon as possible.

III.2 Installation

Nereus is designed to be housed in a standard 19" rack. The unit is 133.35mm high (3U, or 5.25 inches). When choosing a suitable place for installation, please bear the following in mind:

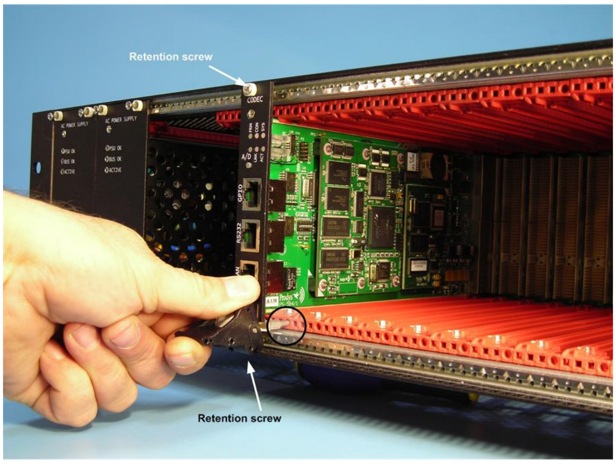
- The position must allow for easy connection of cables.
- The front panel must also be accessible, both for connections and to be able to see LED indicators.
- The air vents must not be obstructed
- We do not recommend that the unit is mounted directly above other equipment, especially ones that generate a lot of heat.

III.3 Inserting and extracting modules

All cards are hot-swappable and can be installed without interrupting the operation of the system. The following instructions are valid for all modules, including the transition cards.

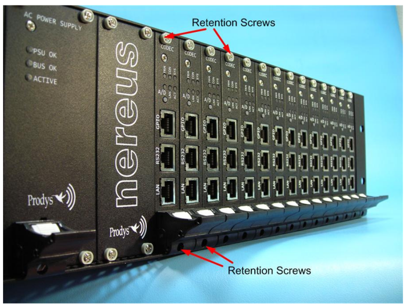
To install a new card follow the next instructions:

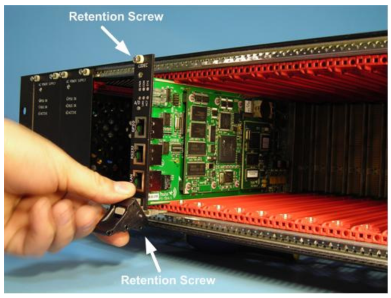
1. Slide the card into the appropriate slot making sure the alignment pin at the bottom of the end plate engage in its socket.
2. Using light finger pressure only, ensure the card is correctly aligned and pressed into the connector.
3. When you are sure that the card is positioned correctly, engage the lever simultaneously. The lever will click to indicate it is locked.
4. While sliding the board, ensure that the card extraction lever is aligned perpendicular to the card flange in the unlocked position and that the board connectors are aligned with the transition card connectors.
5. Locating screws should be fitted to secure the top and bottom of the card.



To extract a card follow the next instructions:

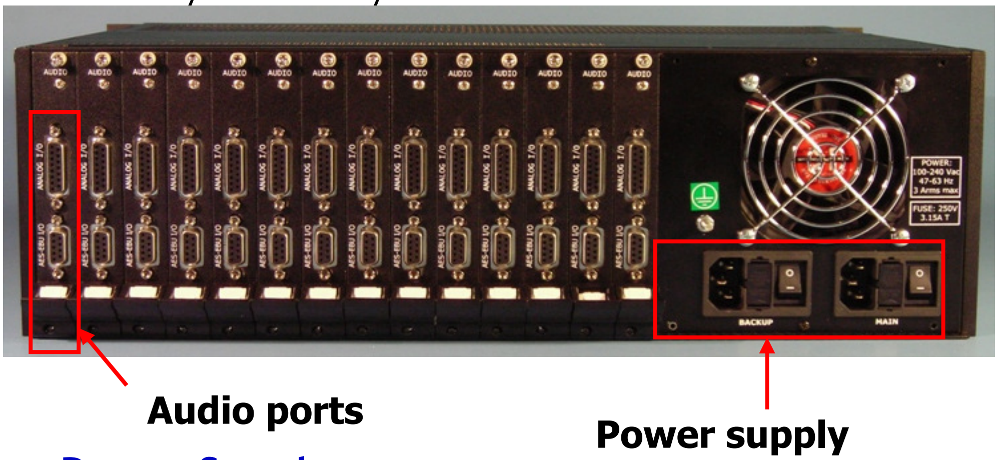
1. Remove the two retention screws located at the top and the bottom of the panel.
2. Press the lock of the lever and slide the card gently.





III.4 The rear panel

The connections of Nereus are spread between the front and rear panel. On the rear panel are found power and audio connections. The power supply is common for the whole system but any audio over IP card has its own audio connections.



III.4.1 Power Supply

On the back panel you will find the main power inlets, one for the main power supply and the other one for the secondary power supply. You will also find the main power switch and the fuse holder for each power input. Nereus is designed to take AC universal power, from 100 to 240 VAC with frequency between 50Hz and 60Hz.

When it is necessary to replace either fuse, it is important to make sure that it complies with the technical specifications outlined below that will ensure adequate protection.

Fuse requirements:

Fuse type:	Type T
Amps	2A
Power	250V

ATTENTION – CHANGING THE FUSE

Disconnect the power cable BEFORE changing the fuse.

WARNING!

HIGH VOLTAGE IS PRESENT WHEN THE UNIT IS PLUGGED IN.

TO PREVENT ELECTRICAL SHOCK, UNPLUG THE POWER CABLE BEFORE SERVICING.

POWER SUPPLY MODULE SHOULD BE SERVICED BY QUALIFIED PERSONNEL ONLY.

III.4.2 Audio interfaces

Each audio over IP card is actually divided in two different cards. One of them is installed from the front panel and the other one from the rear panel. This card contains the analog and digital audio interfaces and so their audio connectors. Analog and digital connections are located in separated connectors: a Sub-D 15 ways connector for analog audio and a Sub-D 9 ways connector for digital audio.

The audio input can be selected from the control software or from the front panel where a switch to select among analog or digital input is located.

Analog or digital outputs are both available at the same time.

The LED above the switch will indicate which input is selected:

- Green colour → Analog input selected.
- Red colour → Digital input selected.

III.4.2.1. Analog audio I/O

The analog audio I/O is connected through the DB15 connector. The wiring conforms to the following scheme:

Pin	Function	Pin	Function
1	NC	9	NC
2	NC	10	GND
3	GND	11	AUDIO OUT RIGHT -
4	AUDIO OUT RIGHT +	12	AUDIO OUT LEFT -
5	AUDIO OUT LEFT +	13	GND
6	AUDIO IN RIGHT -	14	AUDIO IN RIGHT +
7	GND	15	AUDIO IN LEFT -
8	AUDIO IN LEFT +		

These inputs and outputs are electronically balanced with a maximum level of +22 dBu.

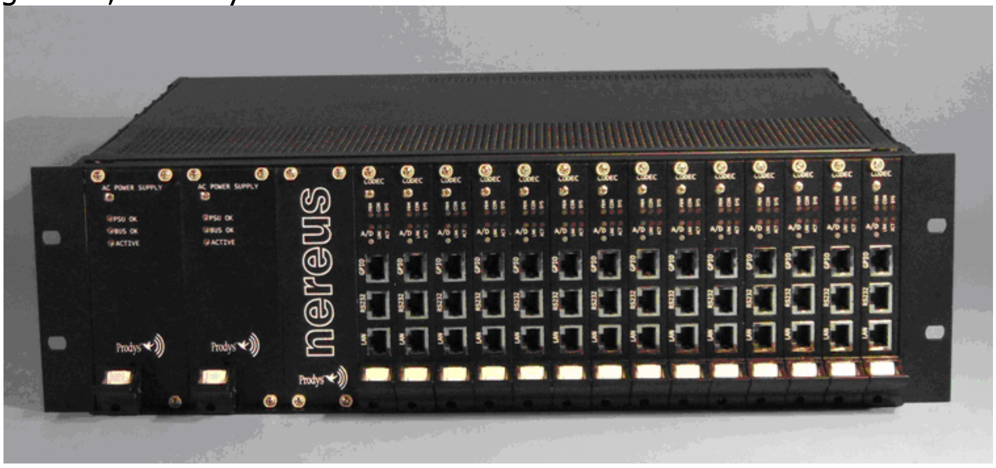
III.4.2.2. AES/EBU Interface

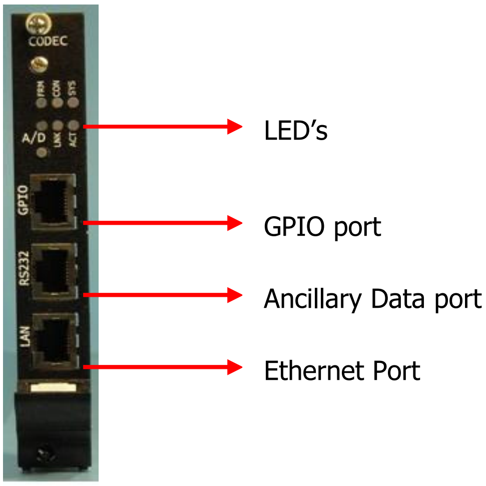
An AES/EBU interface is available via the Sub-D 9 ways connector on the rear panel of each audio over IP card. This connector provides the option to connect an externally synchronised signal. The user can select via software if the digital output is to synchronise with the audio input or with an external sync signal. The connector is wired in the following way:

Pin	Function	Pin	Function
1	AES/EBU IN -	6	AES/EBU IN +
2	GND	7	SYNC +
3	SYNC -	8	GND
4	GND	9	AES/EBU OUT +
5	AES/EBU OUT -		

III.5 The Front panel

On the front panel of each card the communications interfaces are located. Each card has independent connections for each purpose, that is, audio and management, ancillary data and GPIO connections.



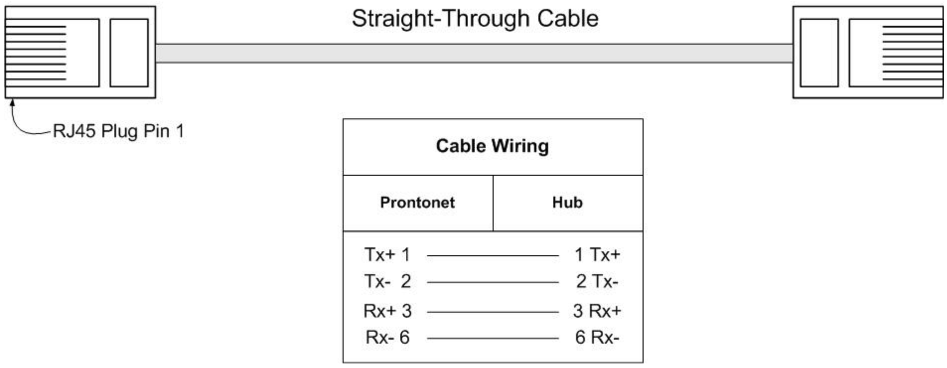


III.5.1.1. Ethernet port - the LAN Connector

The LAN socket is a standard 10/100Base-Tx (10/100 Mbps) Ethernet connection that takes the typical RJ45 plug. Through this Ethernet port it is possible to transmit and receive audio, as well as to control the equipment.

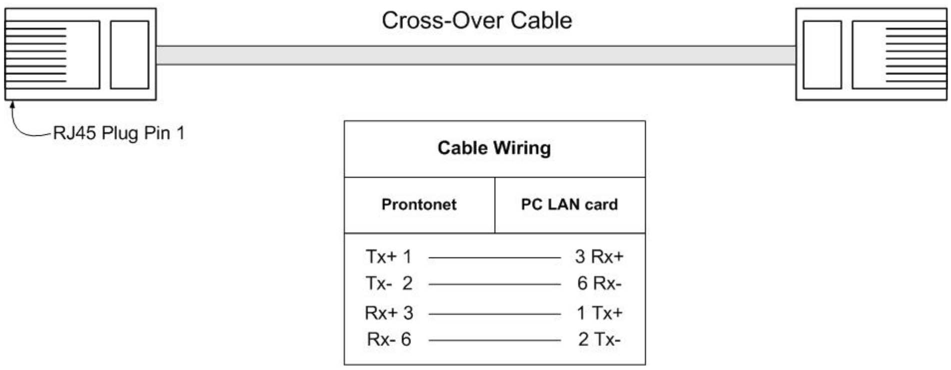
- **Connection to a Hub or Switch**

  In the majority of cases you can simply connect the unit's LAN port to your Ethernet network's Hub or Switch using an Ethernet cable (CAT5). In this case you should use a standard 'straight-through' Ethernet cable (not a 'cross-over' cable). This kind of cable can normally be found in any IT shop. In any case, this cable is described in more detail below:



- **Connection to a PC**

  In some cases, such as when you configure the equipment, it is possible that you will want to connect the unit directly to a PC. In this case the PC must have a free Ethernet port to connect to and you must use a 'cross-over' Ethernet cable. Again, any good IT shop will stock these cables. This time the wiring is as follows:



III.5.1.2. ISDN Port (Optional)

The Nereus ISDN/X21 module incorporates an ISDN terminal adapter that allows connection to a basic ISDN line (2B+D). It supports different ISDN protocols (EURO_ISDN, DMS100, AT&T 5ESS and NAT1). To connect there are two RJ45 connectors: one for connecting to an S/T interface and the other for connecting to a U interface.

Pin	S/T Connector	U Connector
1	NC	NC
2	NC	NC
3	Tx +	NC
4	Rx+	RING
5	Rx-	TIP
6	Tx-	NC
7	NC	NC
8	NC	NC

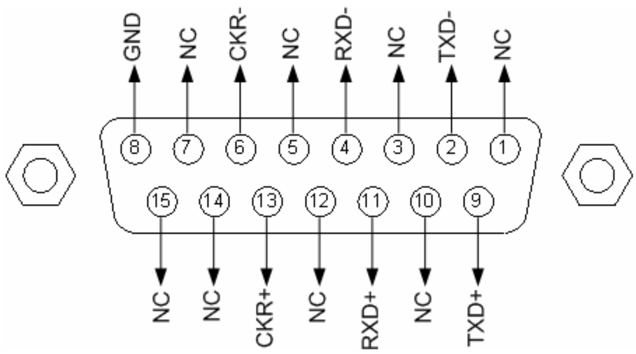
- **The U connector is only available if an NT1 interface is installed.**
- **The NT1 interface is optional and is not supplied as standard.**

When Nereus is connected to a basic rate interface with bus configuration and the unit is the termination point, it must be loaded with 100 Ohm resistors. These may be already fitted in the connection socket, if you do not have external termination, Nereus ISDN module has jumpers available internally that can be set to terminate the ISDN line. The jumpers are found next to the RJ45 connectors.

100Ω RESISTORS CONNECTED

III.5.1.3. X21 Port (Optional)

The X21 Port of the ISDN/X21 Nereus module allows the transmission and reception of audio via a dedicated digital connection. The socket is the standard 15 ways sub-D with the following connections:



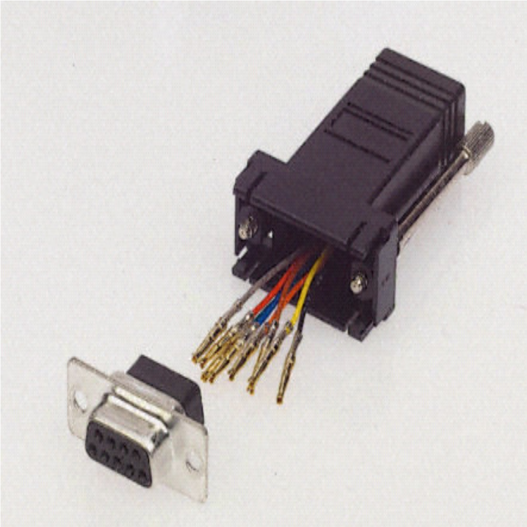
Pin	Function	Pin	Function
1	NC	9	Transmit Data TxD+
2	Transmit Data TxD-	10	NC
3	NC	11	Receive Data RxD+
4	Receive Data RxD-	12	NC
5	NC	13	Clock +
6	Clock -	14	NC (Internally used)
7	NC (Internally used)	15	NC
8	GND		

To connect a V35 port one must bear in mind the following correlation between signals:

Pin	X21 ProntoNet	V35 Signal
2	Transmit Data TxD-	P
9	Transmit Data TxD+	S
4	Receive Data RxD-	R
11	Receive Data RxD+	T
6	Clock -	V
13	Clock+	X

III.5.2 RS 232 Port

The RS232 port is for use as auxiliary data port. This port allows the transmission and reception of data along with encoded audio. Note that this socket is RJ45 connector, as opposed to the typical Sub-D 9 ways connector. To make the conversion between RJ45 and RS232 Sub-D connector there are modular connectors available that should be wired as follows:



Nereus RJ45 Connector	9-pin female D-sub Connector
1 (NC)	1
2 (Rx)	3
3 (GND)	5
4 (NC)	4
5 (NC)	6
6 (GND)	7
7 (Tx)	2
8 (NC)	8

1,4,5,8 must be unconnected

The port is always set to 8 DATA bits, NO parity, 1 START bit and 1 STOP bit. The bit rate can be adjusted to between 300 and 9600 bps via software.

Each card acts as a DCE device, therefore the connection to each of the RS232 ports is wired in the following way:

Nereus audio over IP card – Pin 7 connector RJ45........................Pin 2 PC

Nereus audio over IP card – Pin 2 connector RJ45........................Pin 3 PC

Nereus audio over IP card – Pin 3,6 connector RJ45.....................Pin 5 PC

Hardware handshaking signals are ignored.

III.5.3 GPIO Port

A RJ45 socket provides a general purpose connection with 4 inputs and 4 outputs. The connections must be wired according to the following diagram:

Pin	Function	Pin	Function
1	INPUT 4	5	OUT 4
2	INPUT 3	6	OUT 3
3	INPUT 2	7	OUT 2
4	INPUT 1	8	OUT 1

III.5.3.1. Inputs

The inputs are active for grounding (active low).

Ground is connected to the shield of the RJ45 GPIO connector.

III.5.3.2. Outputs

The outputs are "open collector". They allow an output of 5VDC on one pin to facilitate interconnection with the outputs. Each output supports up to a maximum of 40VDC / 40 mA and will require a pull-up resistor to function with other logic inputs. An appropriate value is 2.2 KOhms.

III.5.4 LED's Description

	Functionality	
FRM	Decoder status	• Green → Decoder framed. • Off → Decoder NOT framed.
CON	Communication status	• Green → Line connected. • Off →
SYS	System status	• Green → Normal operation. • Green blinking → Booting process. • Red → Alarm activated. • Red blinking → Sw updating in process. • Orange → Past alarm.
A/D	Audio input selected	• Green → Analog input selected. • Red → Digital input selected.
LNK	LAN connection status	• Green → LAN connected (physical level detected). Good connection between the card and network. • Red → LAN disconnected. No connection between card and network.
ACT	Rx LAN activity	• Green → Data from the LAN detected. • Off → No data detected.
PSU OK	AC power present	• Green → AC power is available to the Power supply. • Orange → AC power supply is NOT available to the Power supply.
BUS OK	Backplane power supply	• Green → Backplane power supply OK. • Off → Power supply backplane fail.
ACTIVE	Power supply active	• Green → Power Supply operating. • Orange → Power supply as backup.

Chapter V

Remote Control

IV.1 Opening the web browser

Each audio over IP codec card of Nereus can be controlled remotely by using an Internet Explorer web browser connected through the LAN port. The computer can be locally connected directly via a crossover CAT-5 cable, or remotely from a computer connected to the LAN.

To access to each card of Nereus from the Internet Explorer, enter the IP address of the card in the address bar.

Keep in mind that the IP address configuration factory for each card is as follows below and it might be necessary to modify the network configuration of the computer on which the web browser is running.

Slot	IP address default	Slot	IP address defualt
1	192.168.100.100	8	192.168.100.107
2	192.168.100.101	9	192.168.100.108
3	192.168.100.102	10	192.168.100.109
4	192.168.100.103	11	192.168.100.110
5	192.168.100.104	12	192.168.100.111
6	192.168.100.105	13	192.168.100.112
7	192.168.100.106	14	192.168.100.113

Installation Requirements

1.- Pentium 166 or higher.

2.- 64MB RAM minimum.

3.- Operating Systems:

Microsoft Windows XP, Microsoft Windows 2000,

Microsoft Windows NT 4.0 Service Pack 6 or higher,

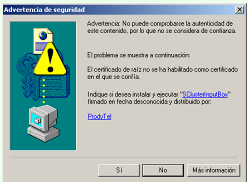
Microsoft Windows Millennium Edition (ME), Microsoft Windows 98.

Microsoft Windows Vista.

4.- Microsoft Internet Explorer 5.0 or higher.

The screen resolution must be 1024x768 minimum.

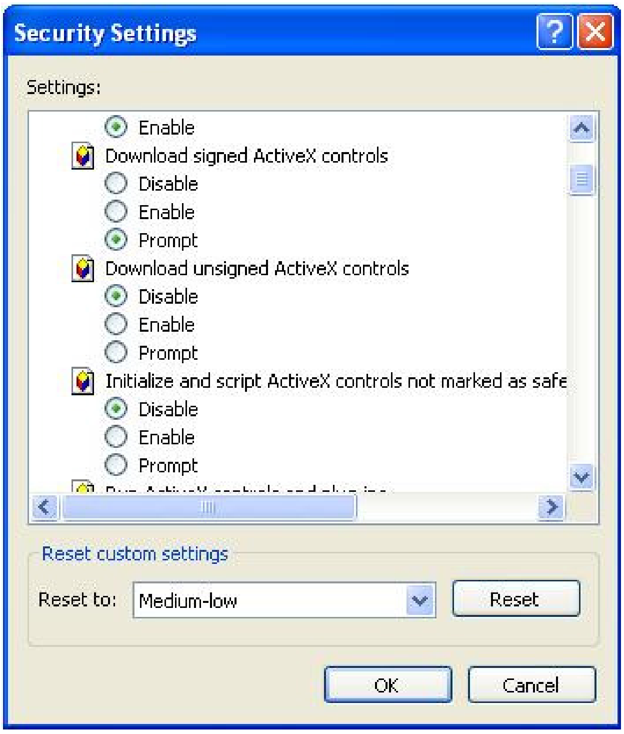
The first time that the computer accesses the Nereus it is necessary to install the software. The computer will show the following window:



Each card is supplied with a different IP address depending on the slot where is installed: 192.168.100.100..192.168.100.113

The first time the user accesses Nereus web page, an OCX file has to be downloaded and installed on the computer. Microsoft Internet Explorer can be configured to block OCX objects installation and/or execution. So, depending on the configuration of the web browser, the following message can appear when first accessing the web page:

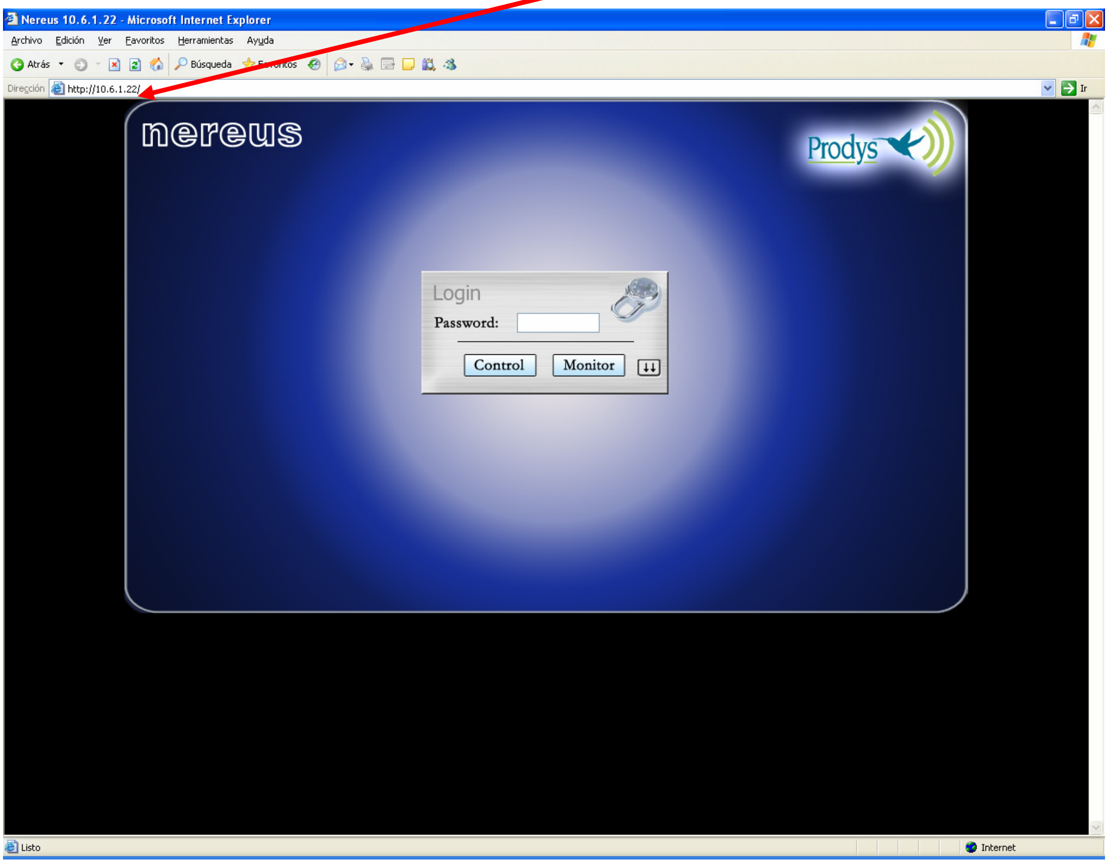
Go to Internet Options in IExplorer, click on 'Security' tab, and set 'prompt' when downloading ActiveX signed and unsigned controls at Local and Internet zones.



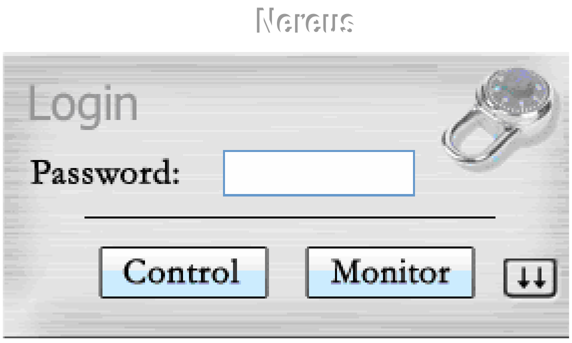
Windows Vista: Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, please disable UAC (User Access Control) on Windows Vista. Once the OCX file has been installed in the computer, UAC can be enabled again.

Each firmware version might have a different OCX file, so the new OCX should be installed as it is done for the first access to the web page of the unit. If the unit was upgraded and, depending on the 'cache' configuration of the Internet explorer, there might be problems when accessing the web page, given that the old web page might be offered by the browser instead of the real one, which should be installed to replace the old one. In this case, a message indicating 'Incorrect Versions' will appear as soon as the user click on 'Control' or 'Monitor' on the login page. Click on F5 to skip the cache entries, and access to the 'real' web page. Even after pressing F5 and, depending on the IExplrorer configuration and/or version, this situation might continue. In that case, go to Internet Options in IExplorer, click on 'General' tab, and delete temporary files.

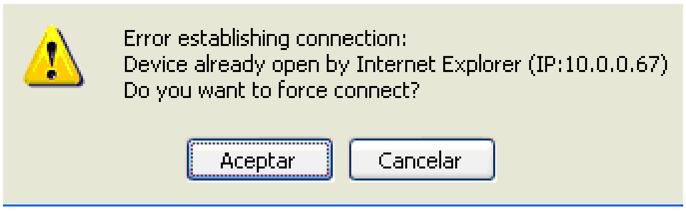
To access to each Nereus card from the Internet Explorer enter the IP address of the unit in the address bar as shown here:



User can choose whether to monitor or to control each card of Nereus from the Web Page. Bear in mind that only one page at the same time can control the unit. However, It is possible to monitor the unit from several web browsers simultaneously.



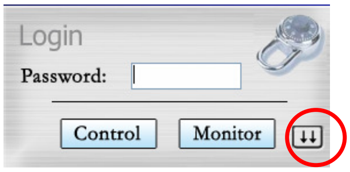
If a unit is already being controlled by a web page and we try to get the control from another website, a message will appear. This message will indicate that the unit is already being controlled from another PC and the IP address of this computer.



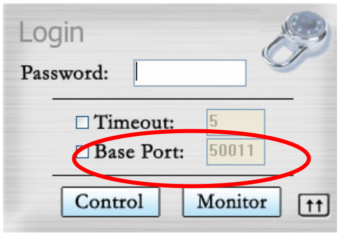
It is possible to get the control by pressing OK. Then, the connection of the old owner will be closed and the unit will be blocked for new controller.

This system is fully integrated with the ProdysControl. In that way, the control from/to ProdysControl can be revoked. It is possible to have a unit being controlled by ProdysControl and at the same time, web browsers monitoring the same unit, etc…

When entering the web page, the user can set the time-out period for the connection between the PC running the web browser and the unit[1]. By default, this period is set to 5 seconds. To modify this parameter, click on the advance features button of the login window:



and specify the new value from the following window:



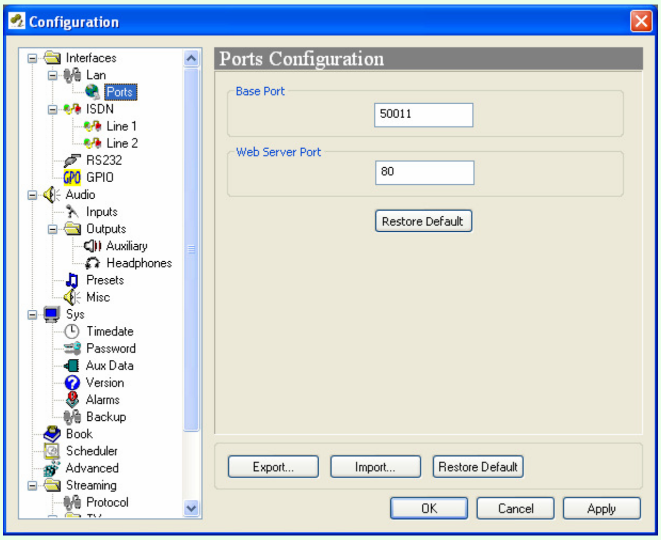
The user should take into account that the longer the time-out, the less likely it is that the connection between the PC and the unit will be lost.

It is very important to differentiate between this time-out value, and the one for audio streaming connections.

Base Port: When Nereus ports have been modified, it will be necessary to change this parameter according to the new port configuration in order to get access to the Nereus web browser.

MORE ABOUT THE PRODYS PORTS
Changing Prodys Ports: The Nereus configuration menu allows the user to configure which ports the unit will use for its TCP/UDP/IP communications from the web page.

[1] This option is available from version 4.8.1 onwards.



There are two different groups:

- **Web Server Port**: By default, it is TCP port 80. This is the internal web server port.
- **Base Port**: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit. From this base port on, up to 30 ports should opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required).

IMPORTANT

The following should be taken into consideration when changing the base port:

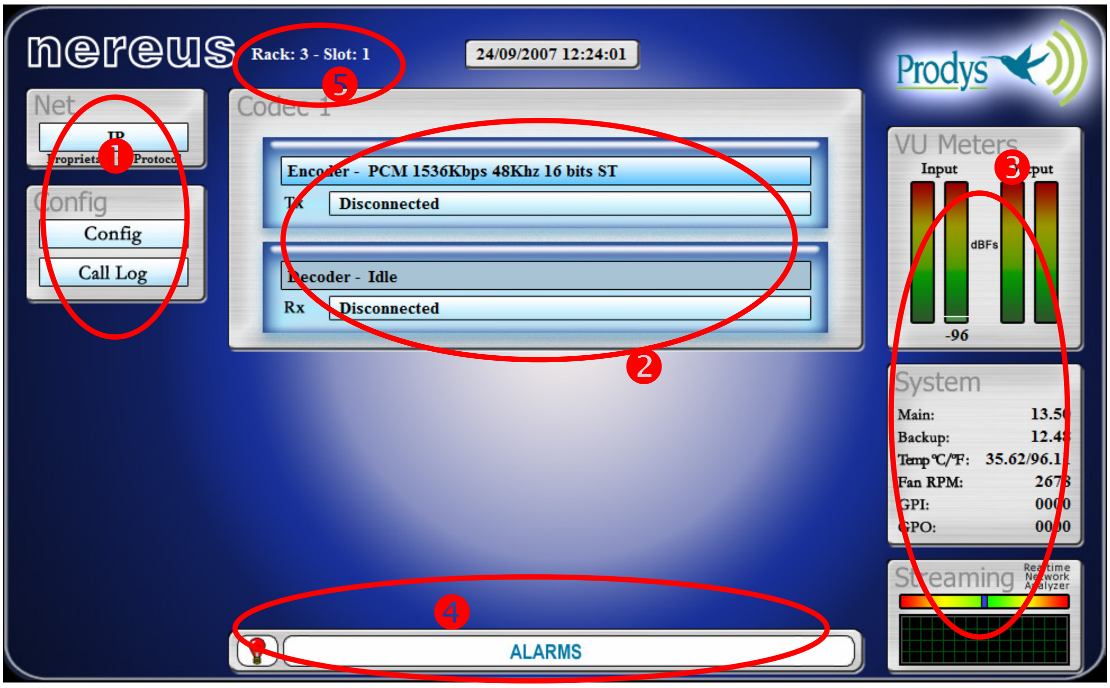
- **To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon: http://<IP>:<Port> Example: 192.168.0.10:8080**

IV.2 Main Screen

Once the page is entered correctly, the web browser will display the "Home Page". The home page of the Web page is arranged in five main areas:

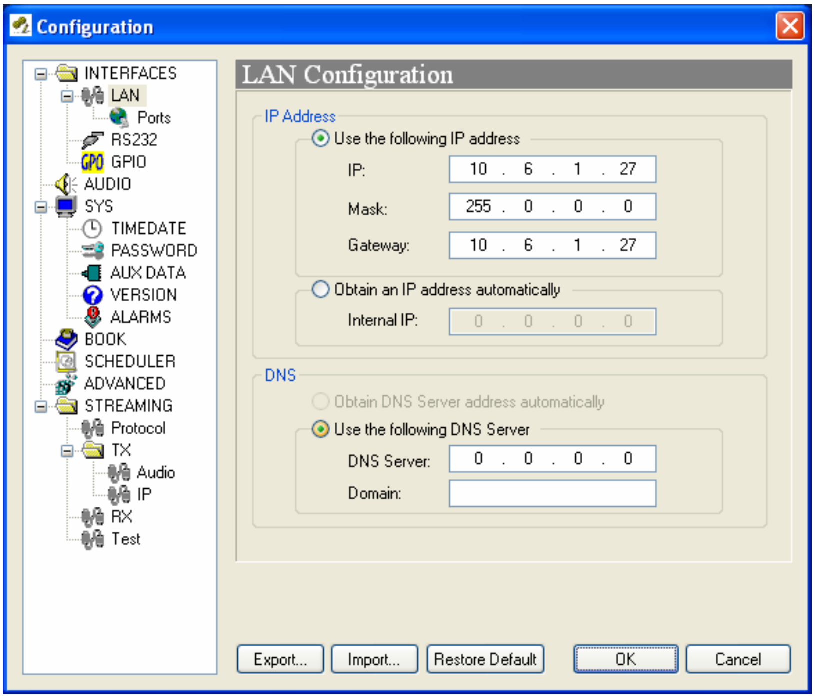
- General Configuration area. ❶
- Control area. ❷
- Monitor area. ❸
- Alarms Window. ❹
- Nereus identifier and slot location. ❺

By clicking on any button on the menu bar or any highlighted zone, individual configuration pages are displayed.



IV.3 Configuration Options

When the Conf button is clicked the configuration menu is displayed:



IV.4 Saving and Importing Configurations

It is possible to save the configuration in a file by using the “Export” utility option. When this option is clicked the saving dialog is open and the user must select the file where the configuration will be stored.

To restore a configuration the “import” option must be clicked and then a file configuration must be selected.

From version 4.5.2 onwards, the factory default configuration can be restored from the web page. This option allows the user to restore the factory default configuration, except those parameters related to the LAN configuration: IP address, netmask and gateway.

To apply the changes, the user can press either the OK button or the Apply button[2]. By pressing the Apply button the changes will be applied but the configuration window will remain.

[2] Apply button available from version 5.0.0 onwards.

IV.5 Configuring Interfaces

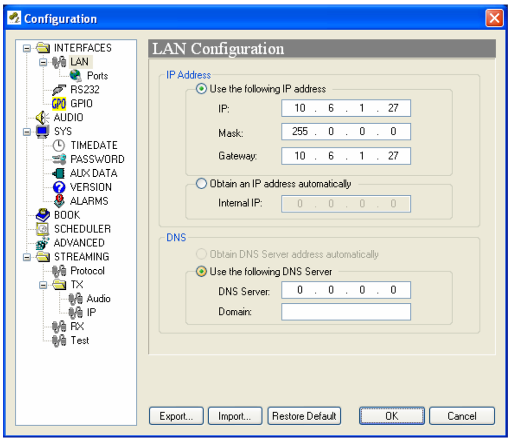
By clicking on the INTERFACES icon the port configuration dialog appears. In the left side the ports can be selected. The right window shows the dialog to configure the selected port. For example, if we select the LAN port:

IV.5.1 LAN

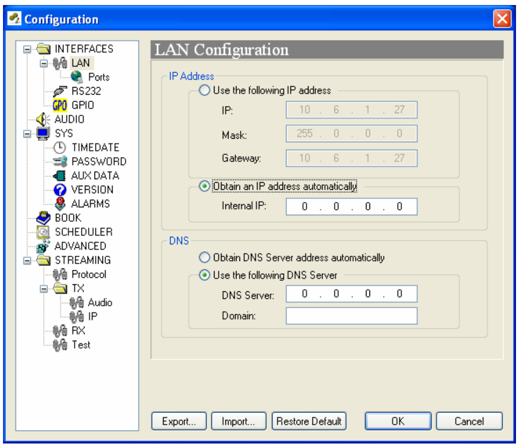
IV.5.1.1. Assigning an IP address

Nereus allows the user to assign the IP parameters both manually and automatically (DHCP).

- Manually → User must enter the following IP parameters manually: IP address, Mask and Gateway IP address.



- Automatically → Check "Obtain an IP address automatically" option to get the configuration automatically from a DHCP Server.



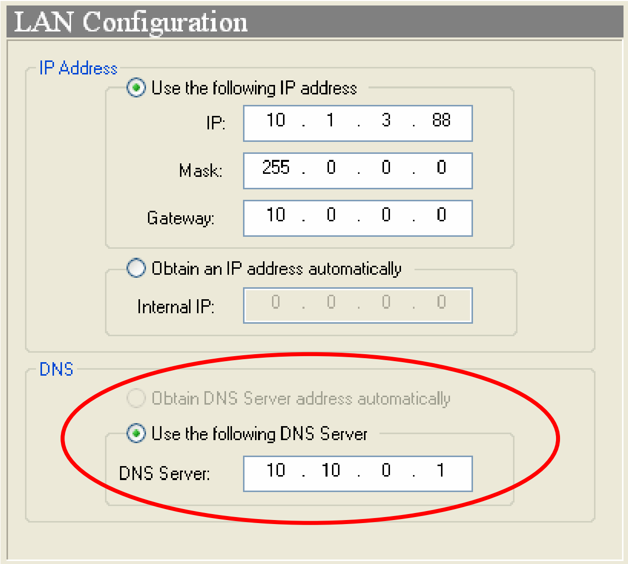
By setting DHCP, the unit will receive its IP parameters when starting. These IP settings might be different from time to time, that is why Nereus supports RIP2 protocol. This protocol allows the user to set an 'internal' IP address, in order that the unit can be identified regardless of the IP settings provided by the DHCP server.

IV.5.1.2. DNS configuration

DNS (Domain Name Server) allows the user to use names instead of IP address. A DNS server mantains a data base of domain names (host names) and their corresponding IP addresses.

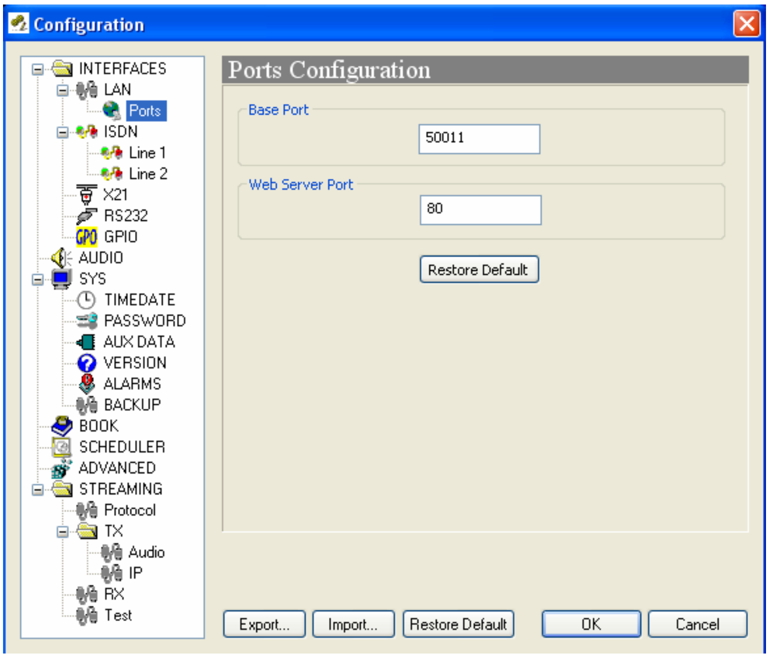
Once the DNS server address is introduced, the host name can be used for all operations that require to write an IP address (dial, phone book, SNTP configuration, etc).

The DNS server IP address can be introduced manually or obtained automatically by means of DHCP.



IV.5.1.3. Ports

It is possible to configure which ports the unit will use when selecting Prodys proprietary protocols for IP communications[3].



There are two different groups:

1. **Web Server Port**: By default, it is TCP port 80. This is the internal web server port.

2. **Base Port**: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit. From this base port, up to 30 ports should be opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required).

The following things should be taken into consideration when changing the base port:

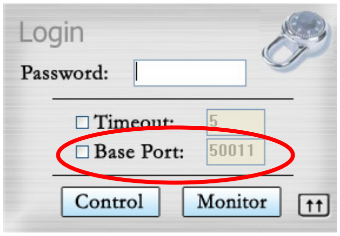
1. To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:

   http://<IP>:<Port> Example: 192.168.0.10:8080

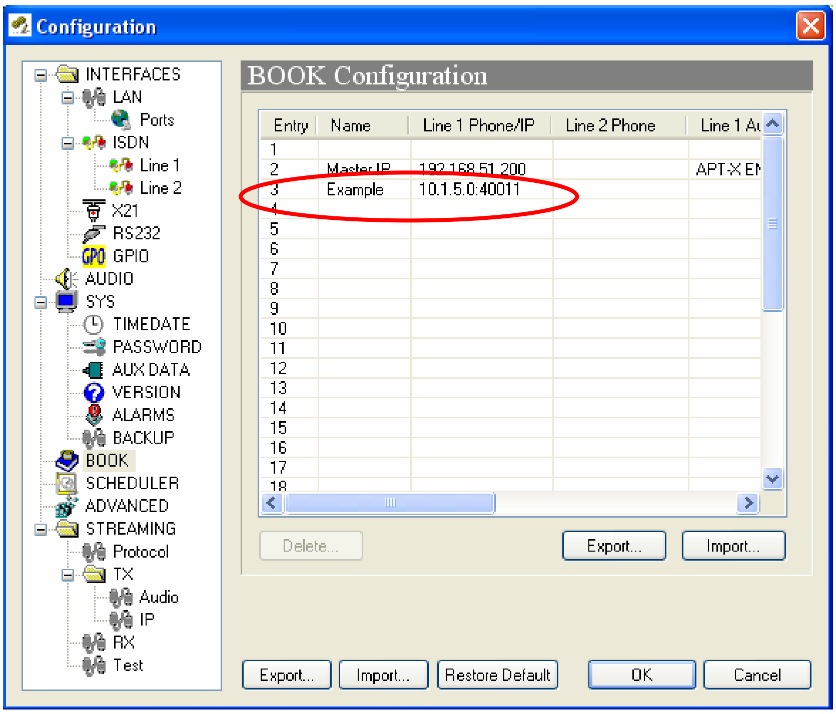
[3] For more information, please read chapter V.6.5 – Prodys proprietary set of Protocols.

To log into the unit, click on the advance features button of the login window:

and specify the base port which we want to connect to:

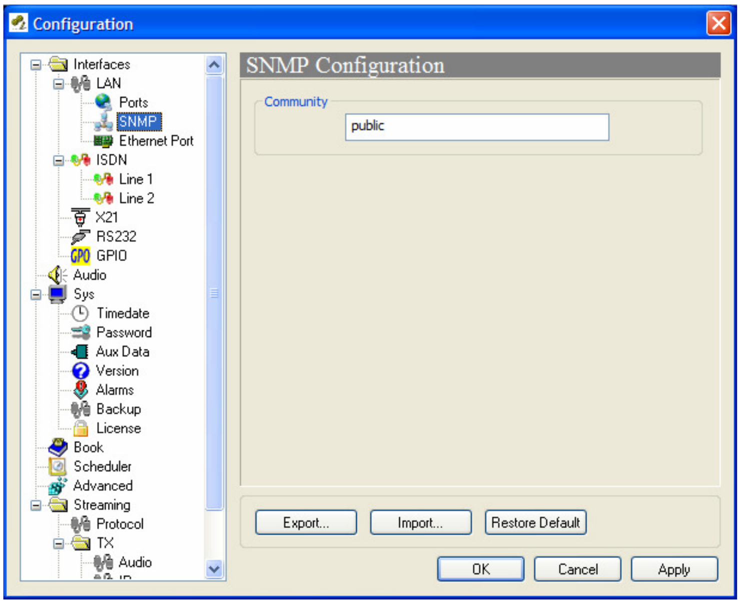


2. When establishing a call, the new base port must be entered after the IP address, separated by a colon. Example: When calling to a unit which IP address is 192.168.1.2, and base port 40011, the following call destination should be entered into the 'Dial' window: **192.168.1.2:40011**.
3. When establishing a bidirectional call with Prodys IP codecs, two connections are made automatically, one for each direction. For the receiver to call properly to the caller (*only in case the base port in the caller is not the default one*), it should have an entry on its phone book with the IP address of the caller and the base port of the caller. Example: if the caller has the IP address 10.1.5.0 and its base port is 40011, in the receiver, the following entry should be configured in its phone book:



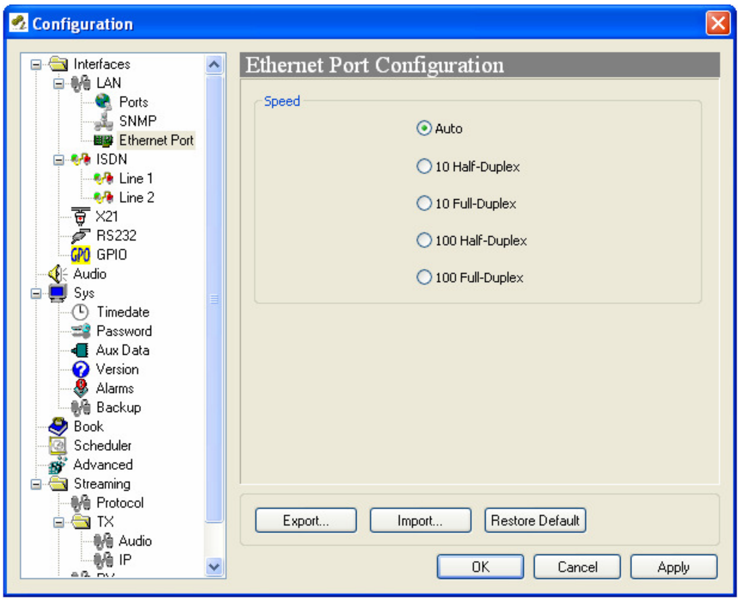
SNMP configuration[4]

It is possible to set the SNMP community.



Ethernet Port Configuration[5]

It is possible to set speed and duplex configuration to the following values:

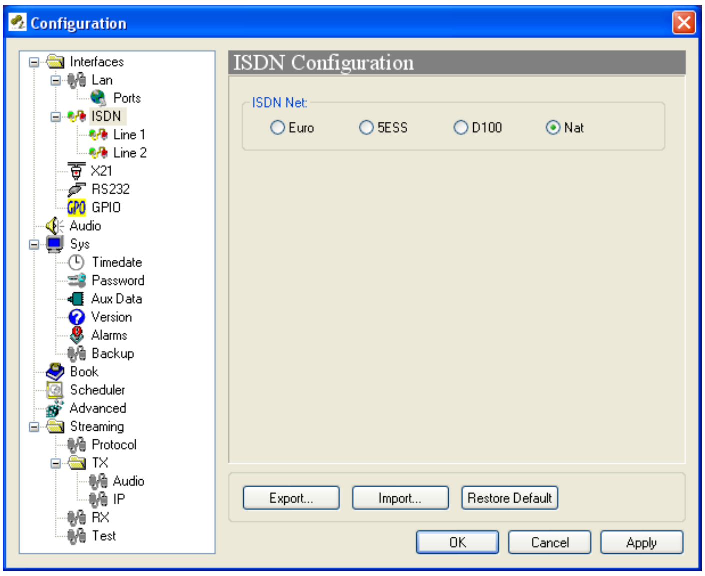


[4] This option is available from version 5.2.1 onwards.

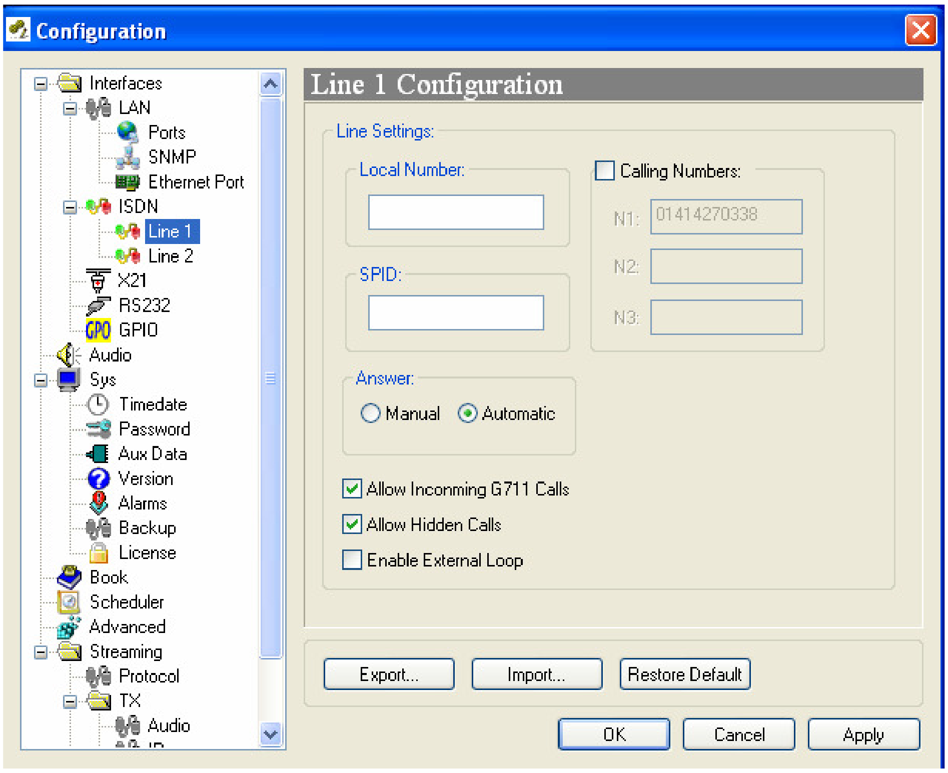
[5] This option is available from version 5.2.1 onwards.

IV.5.1.4. ISDN Terminal adaptor Configuration (Optional module)

The first dialog allows the selection of the ISDN protocol. There are four options: Euro-ISDN, 5ESS, DMS 100 and NAT1:



The options Line 1 and Line 2 allows the user to configure in each line (B channel) the Local Number, Calling number filters and the answering mode:



- **Automatic or manual response:** Incoming calls can be answered automatically or manually depending how the menu option ANS is

configured for the ISDN port. If the ANS mode is set to manual, the appropriate CALL key must be pressed to accept the call and connect to it.

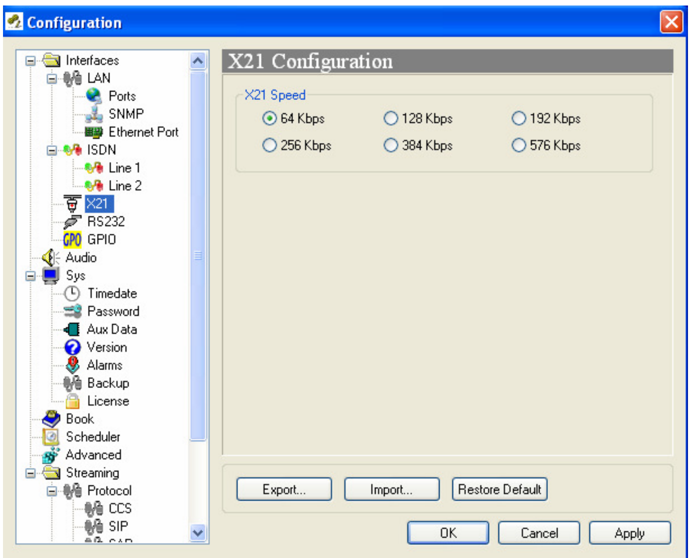
- **Call filters:** It is possible to record up to three numbers for each line that work as call filters, meaning that the line will only connect to calls that come from these pre-programmed numbers. This option is found in the ISDN set up menus under CNUM (Calling Number).

- **Local number**: It is also possible to assign a single number to each line in a way that the line will only respond to calls to this local number. This can be used if you need to map an ISDN directory number to a specific audio port. This option is found in the ISDN set up menus under LNUM (Local Number).

  - Enabling/disabling G711 ISDN incoming calls:
    From version 4.6.0 onwards, a new check box allows the user to enable/disable voice calls (G711) over ISDN.
  - Enabling/Disabling hidden-number ISDN calls:
    From version 4.6.0 onwards, a new check box allows the user to enable/reject incoming ISDN calls with undefined number.
  - External ISDN loop[6]:
    It is possible to set a loop for the ISDN interface, so that data coming from any ISDN connection will be sent back. This functionality works over both B channels.

IV.5.1.5. X21 Port (Optional module)

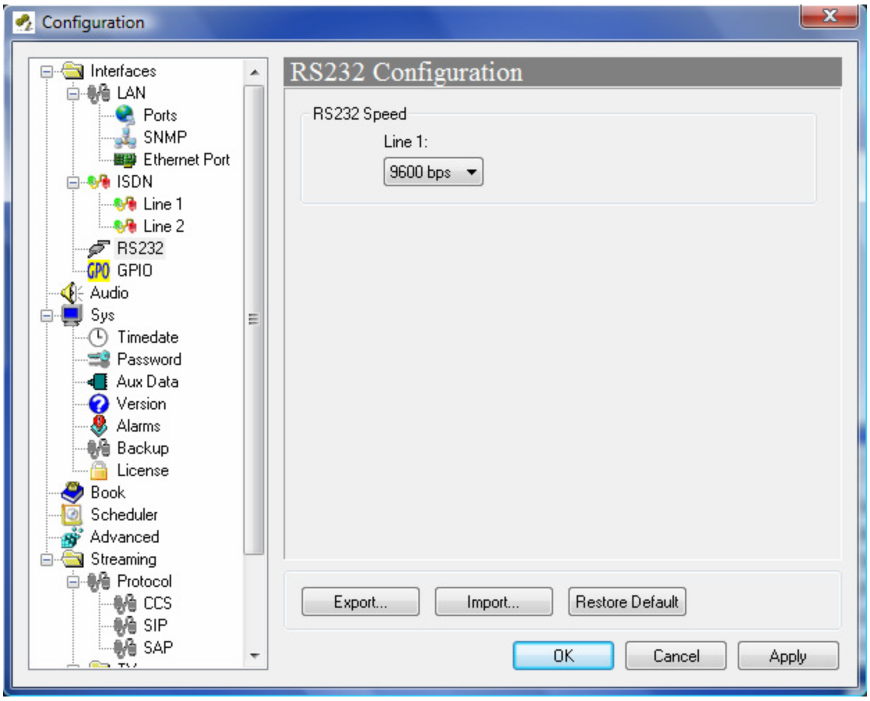
From this option the speed of the X21 port is selected.



[6] This option is available from version 5.2.1 onwards.

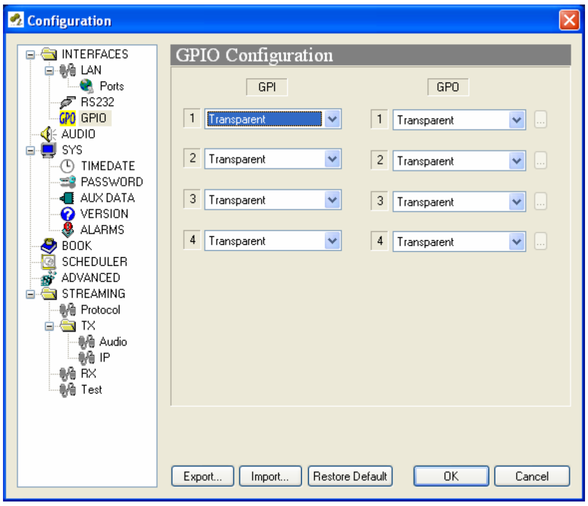
IV.5.2 RS232 Port

There is one RS232 port for use as auxiliary data port. This port allows the transmission and reception of data along with encoded audio. The ports are always set to 8 DATA bits, NO parity, 1 START bit and 1 STOP bit. The bit rate can be adjusted to between 300 and 9600 bps via software.



IV.5.3 GPIO Port

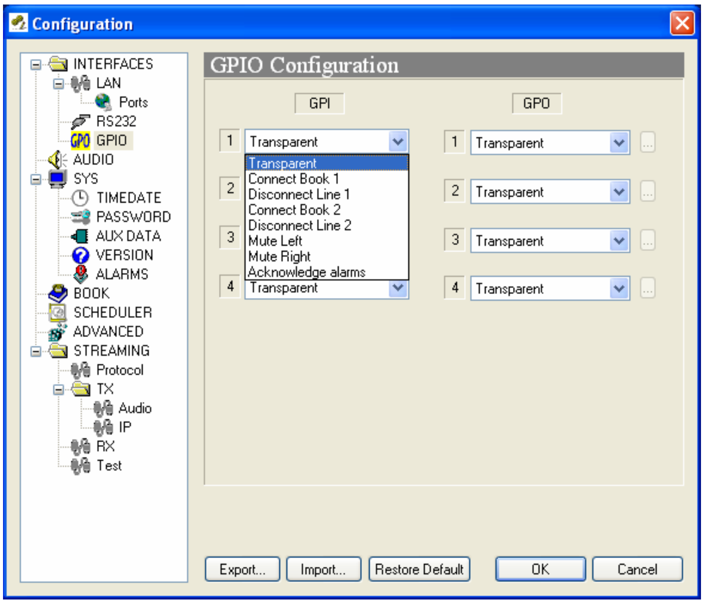
From this option the inputs and outputs of the GPIO port are configured.



IV.5.3.1. Inputs

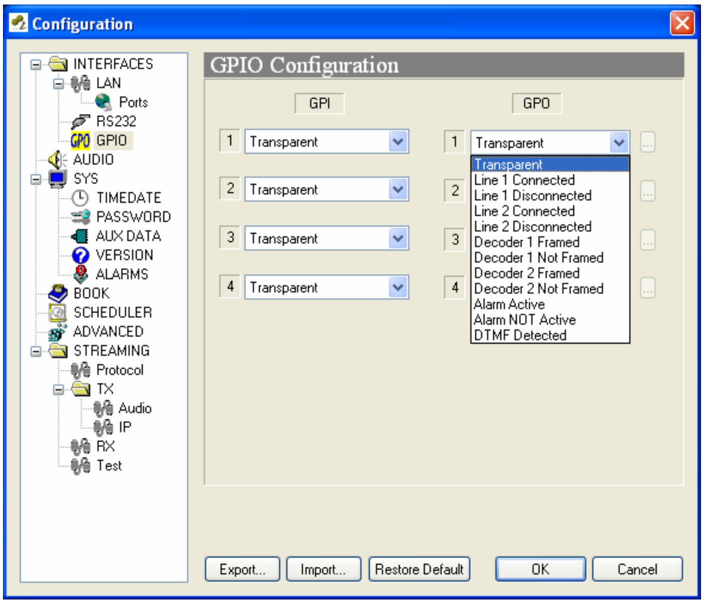
INPUTS	
Transparent	Under this configuration, the state of the input will be present in its homologous output in the Nereus card connected in the other end.
Connect Line 1	When this input is activated, Nereus card will proceed automatically to connect the line 1.
Disconnect Line 1	When this input is activated, Nereus card will proceed automatically to disconnect the line 1.
Connect Line 2	When this input is activated, Nereus card will proceed automatically to connect the line 2. It makes sense only when Nereus card is working in "Double Mode" (see System Configuration – IP Codec Mode).
Disconnect Line 2	When this input is activated, Nereus card will proceed automatically to disconnect the line 2
Mute Left	When this input is activated, the left audio output will be muted.
Mute Right	When this input is activated, the right audio output will be muted.
Acknowledge alarms	When this input is activated, the alarms will be acknowledged.

1. **The inputs are activated by grounding**

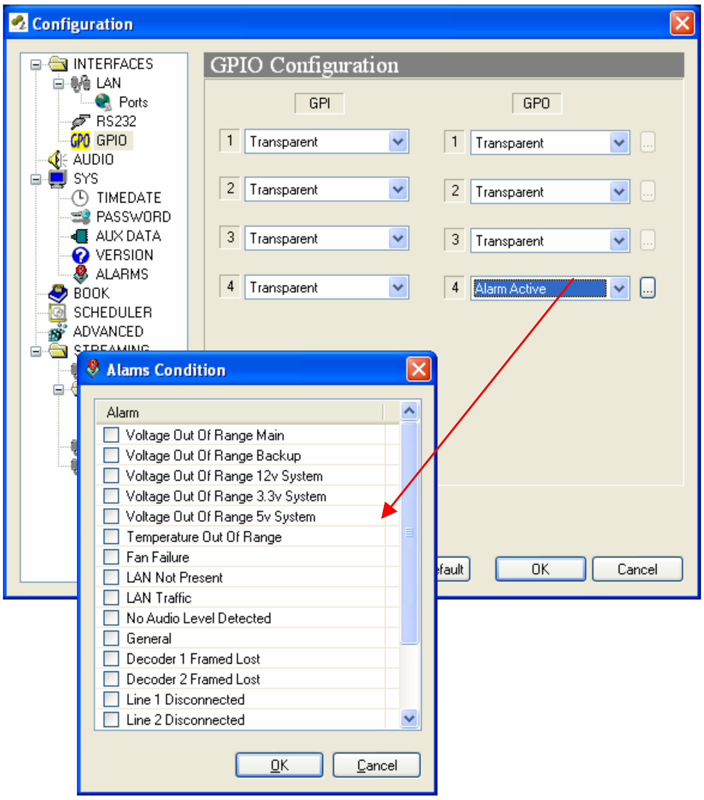


IV.5.3.2. Outputs

OUTPUTS	
Transparent	Under this configuration, the state of the output will be the same that its homologous input has in the Nereus card connected in the other end.
Line 1 Connected	The output will be activated when the line 1 is connected.
Line 1 Disconnected	The output will be activated when the line 1 is disconnected.
Line 2 Connected	The output will be activated when the line 2 is connected. It makes sense only when the Nereus card is working in ISDN mode as a DUAL CODEC.
Line 2 Disconnected	The output will be activated when the line 2 is disconnected. It makes sense only when the Nereus card is working in "Double Mode" (see System Configuration – IP Codec Mode).
Decoder 1 Framed	The output will be activated when the Decoder 1 is Framed.
Decoder 1 NOT framed	The output will be activated when the Decoder 1 is NOT Framed.
Decoder 2 Framed	The output will be activated when the Decoder 2 is Framed. It makes sense only when the Nereus card is working in ISDN mode as a DUAL CODEC.
Decoder 2 NOT framed	The output will be activated when the Decoder 2 is NOT Framed. It makes sense only when the Nereus card is working in "Double Mode" (see System Configuration – IP Codec Mode).
Alarm Active	The output will be activated when one Alarm is activated.
Alarm NOT Active	The output will be activated when there are NOT Alarms activated.
DTMF detected	DTMF is detected by the Decoder.
Line 1/2 Status	Line Disconnected: GPO set to '0'. Line Connected: GPO set to '1'. Trying/receiving a call: GPO blinking (1 second period).



When a GPO is configured to monitor alarms ("Alarm active"), it is possible to select which alarms will enable this output. This GPO will be activated when one or more of the selected alarms arise.



Temperature out of range alarm will arise when the temperature goes over 50ºC. Lan traffic alarm will come up when more than 90% of the LAN bandwidth is in use.

This GPO will be activated when one or more of the selected alarms arise.

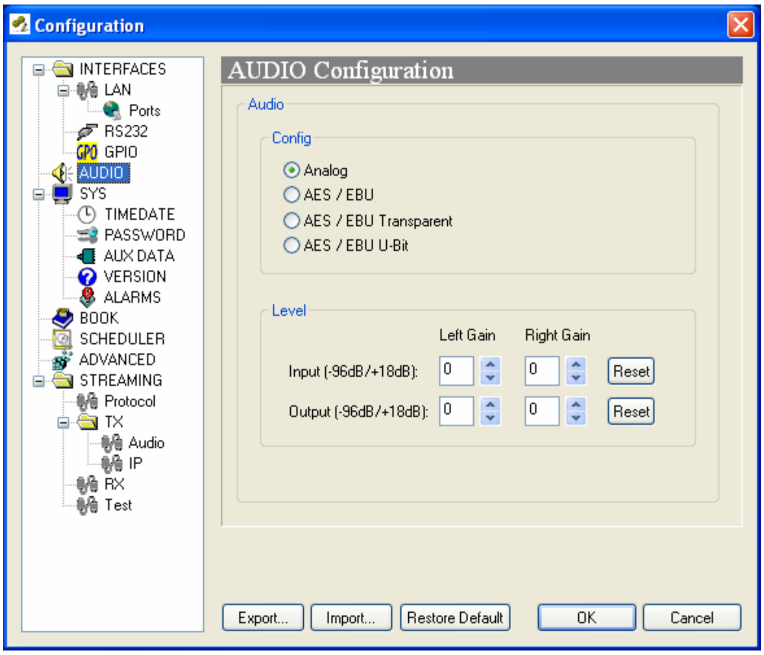
In addition, it is possible to program a GPO to be enabled when a DTMF is detected in either Decoder. The decoder number and the DTMF which will enable the GPO must be defined by the user[7].

[7] From version 4.5.0 onwards.

IV.6 Audio

From here select the audio input (analog or digital). If it is digital we can choose between synchronising with the audio input or with an external clock.

Audio gain adjustment: Audio inputs and outputs can be adjusted, even when lines are connected, and each audio channel independently.



It is possible to select Digital Audio Output synchronization to 'external' when the audio input is set to analog[8].

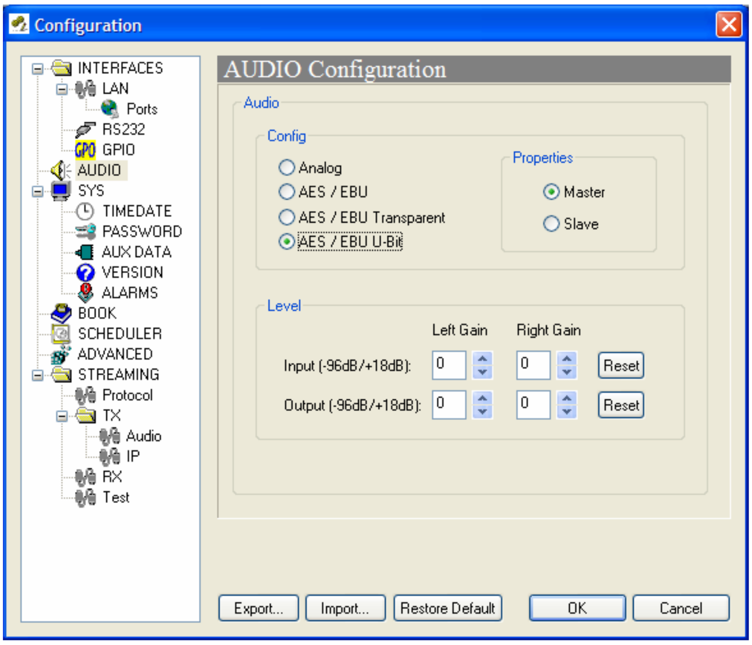
IV.6.1 AES/EBU transparent

The AES/EBU Transparent is a special transmission mode for PCM 24 bits digital audio without any processing at all. This mode allows the transmission of "NON-PCM audio" or data between devices that use AES3 digital audio interface, for example Dolby E systems.

IV.6.2 AES/EBU U-bit

This new feature permits the transmission and reception of the U-Bit of the AES/EBU audio frame via IP, regardless of the compression mode used for the audio connection.

[8] From version 5.2.1 onwards.



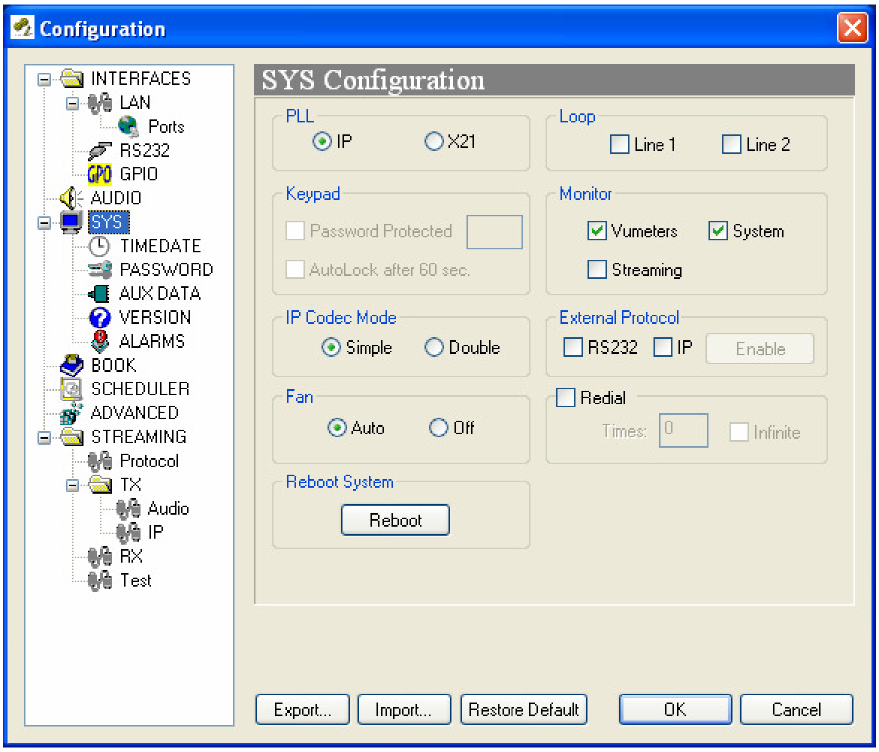
For this feature to work properly, one of the units be configured as master, and the other one as slave.

In addition, the following restrictions apply:

1. **The minimum valid TBP (Time Between Packets) is 4 msc. Less than that won't work properly.**
2. **For it to work, the master unit has to be caller, and the slave one the callee.**

IV.7 System Configuration

Grouped here are functions that affect the general operation of the unit:



IV.7.1 PLL

This is a special option that allows you to select the PLL reference clock when NET = IP.

IV.7.2 Loop

This sets up an Encoder-Decoder loop. Its purpose is to help the user to find problems with audio connections. The loop takes place in accordance with the configuration available at that moment.

IV.7.3 Monitor

This option allows the user to hide or display additional information on the main window:

- VU Meters for audio inputs and outputs.
- System: Monitor voltages, temperatures, and fan operation.
- Streaming: Shows clock-sync algorithm operation and Buffer Occupation Graph (Real Time Network Analyzer)[9]. This algorithm comes into scene when receiving audio over IP. This display is active when the audio data is "framed" - Under

[9] For more information please read chapter V.6.5 – Prodys Proprietary set of Protocols.

normal circumstances the indicator should be in the middle Green area.

IV.7.4 Reboot System

A new option to reset remotely a unit was included in the SYS configuration menu[10].

IV.7.5 Nereus external protocol

The ProntoNet Family Software Development Kit provides a tool to manage Prodys units from external applications or devices different than ProdysControl. In this manner, it is possible to customize the control of the units and to integrate them in a global management system.

The protocol is based on which can be managed through two different communication ports: the LAN port and the RS232 port. For more information, please read ProntoNet SDK User's Manual.

IV.7.6 IP Codec Mode

The normal operation is work with one streaming communication. This is what is called "Simple Mode". However, it is possible to enable a second streaming communications that can be used with some restrictions. When the "double mode" is selected, a second streaming communication is possible and the main screen is modified to show the control of this second communication.

[10] Available from version 4.7.1 onwards.

Restrictions in "Double Mode"

1.- Encoder 1 and Encoder 2 can be configured in MONO and not in Dual or Stereo.

2.- Encoder 2 only works in G722.

IV.7.7 Redial

It is possible to configure a number of redials. Nereus will try the connection in the following cases:

1.- The unit is trying the connection for the first time.

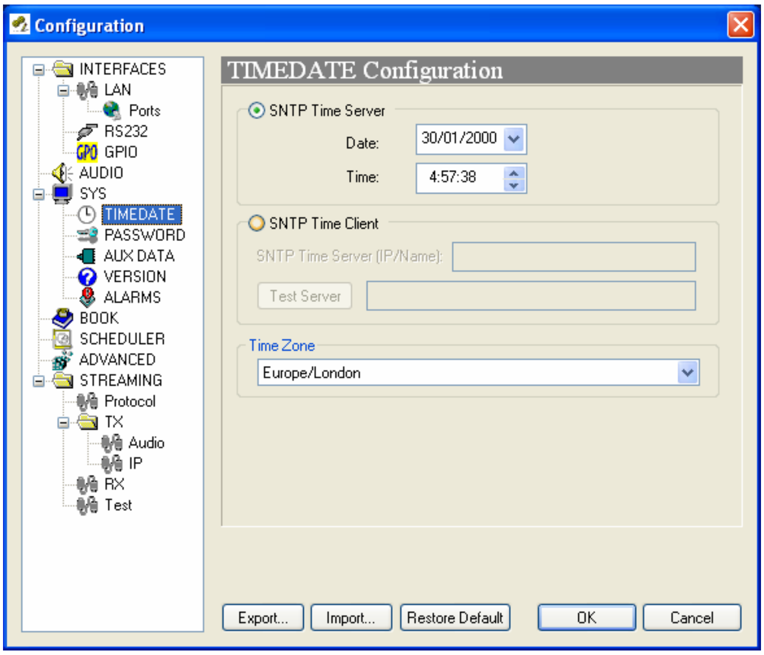
2.- The unit was connected and lost the connection because the line dropped or the line was disconnected from the other end.

IV.7.8 Reboot System

This option allows the user to reset remotely the unit.

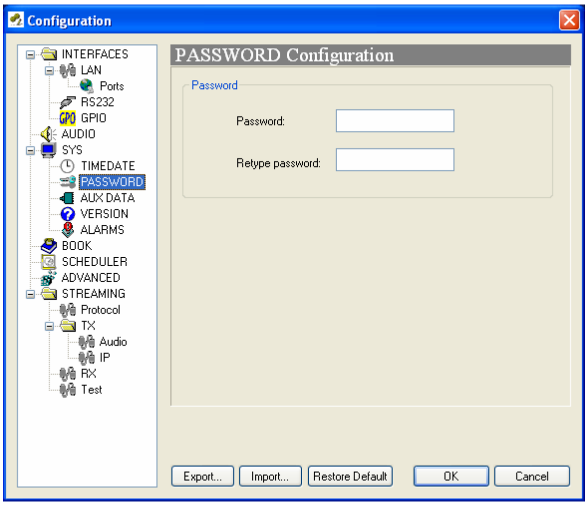
IV.7.9 Time-Date

This option allows synchronizing time and date by using the SNTP protocol (Simple Network Time Protocol). SNTP operates always in the client-server model and for this reason, Nereus can work as SNTP server or SNTP client.



IV.7.10 Password

Whenever the web browser is started, it is necessary to enter a security password. This is the dialog to program it.



IV.7.11 Aux Data

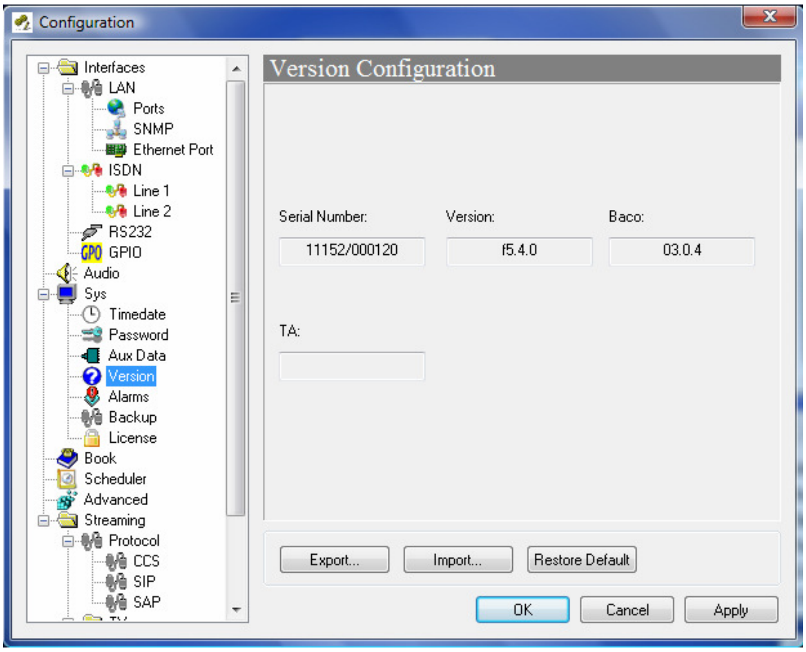
- RS232: There is one RS232 port for use as auxiliary data port. This port allows the transmission and reception of data along with the encoded audio. When one option is checked, the corresponding decoder will allow the reception of ancillary data.

The auxiliary data can be sent via a different path, different from the audio one (Aux. data embedded in audio). This method has 3 big advantages: less delay independent from the audio codification delay; the possibility to send/receive auxiliary data regardless of the compression mode used for the audio communication; and the possibility to send/receive the User bit of the AES/EBU frame. The drawback is that the audio and data delay won't be the same[11].

- GPO: Under 'transparent' configuration, the state of the input will be present in its homologous output in the unit connected at the other end.

[11] Before version 4.8.1, auxiliary data could only be sent embedded into the audio frame, for those compression modes which accept it. From version 4.8.1 to version 5.2.1, the auxiliary data could only be sent on a different IP path, and from version 5.2.1 onwards, it is possible to select between both ways of operation.

IV.7.12 Software Versions:



IV.7.12.1. Backup

Backup Configuration. Nereus allows the user to use the ISDN line as a backup when the unit works in IP or X21 mode.

The backup is enabled by clicking the "Backup" box.

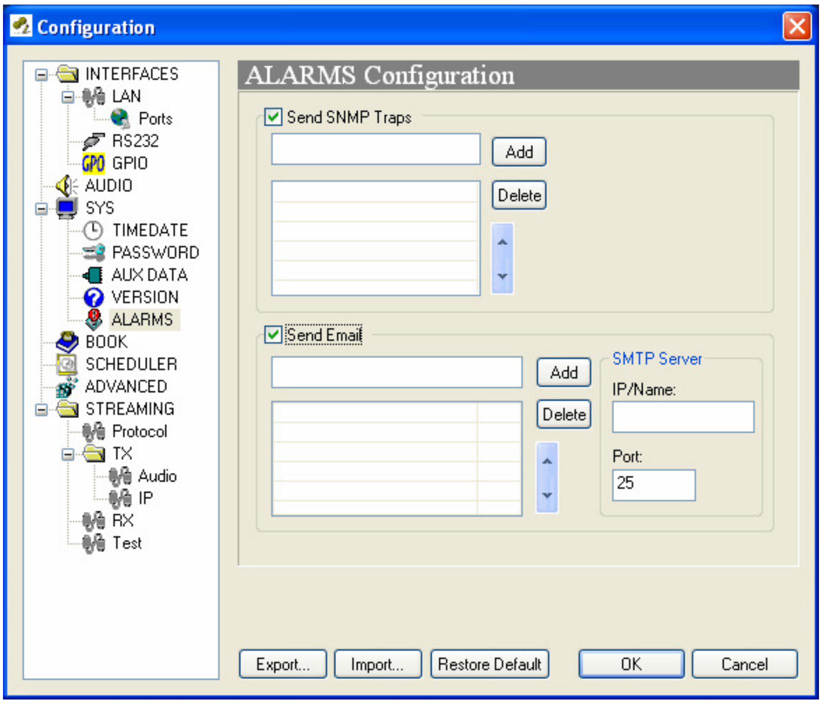
For more information about the backup operation see chapter V.11 – How the backup mode Works?

IV.7.13 Alarms configuration

This menu allows the user to activate two different ways to report remotely the alarms.

IV.7.13.1. SNMP Traps

From this option it is possible to select the IP address where the SNMP traps will be sent. An SNMP manager located at this IP address will receive and process the information according to the SNMP protocol. These SNMP traps will notify alarm information, that is, an SNMP trap will be sent when an alarm is activated or deactivated, with the time and type of alarm. For this to happen, alarms have to be enabled (see chapter IV.16 - Alarms). Nereus can be fully monitored though SNMP protocol.



IV.7.13.2. Send an e-mail

A second option is to send an e-mail when an alarm is activated. The dialog allows introducing one or more email addresses where the alarm report will be sent.

Additionally it is possible to introduce the IP address of a SMTP server which will allow the transmission of a SMS reporting the alarms.

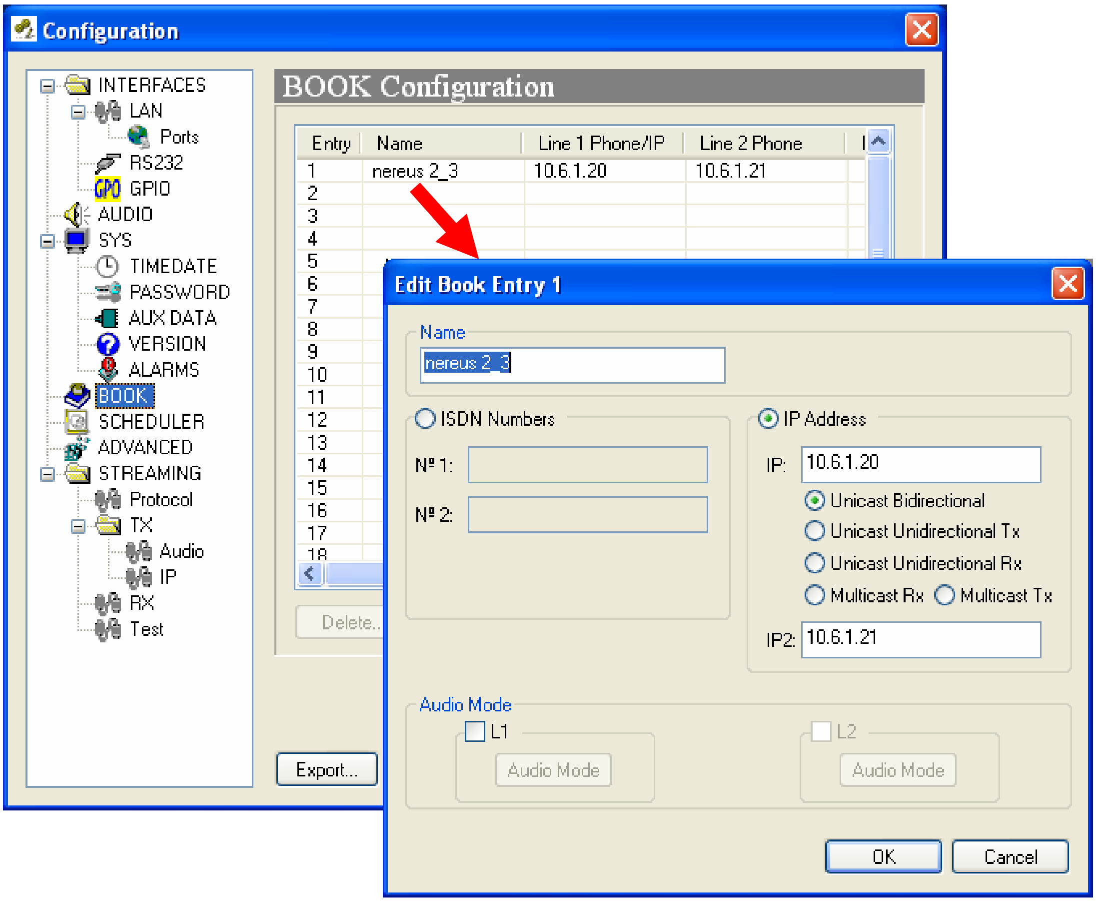
IV.8 BOOK

When the BOOK option is selected, the phone book window is showed on the right side. The Phone Book records the user name, an IP address and the audio mode. There are up to 64 indexes[12].

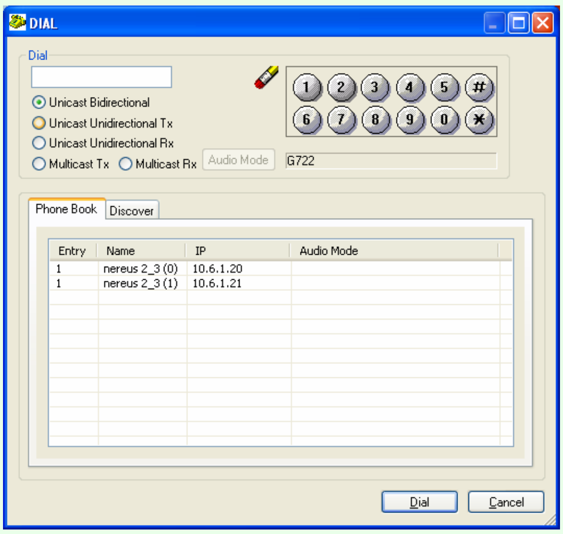
The procedure to edit a phone book index is as follows:

1. To select one index by clicking on its area.
2. To select IP Address.
3. To enter the IP address.
4. To configure the encoder is optional. In case of that it is not entered, the unit will proceed to call in the current encoder configuration.
5. The encoder configuration for line 2 is disabled because it always works in G722.

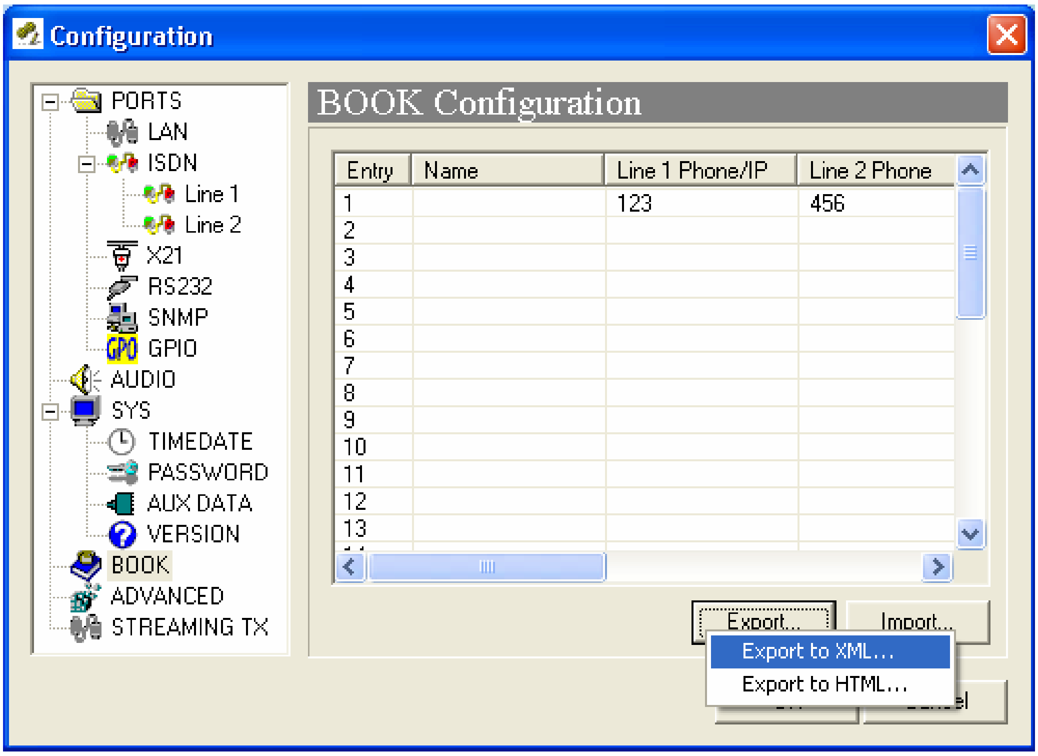
[12] Before version 5.2.1, the number of available entries was 32.

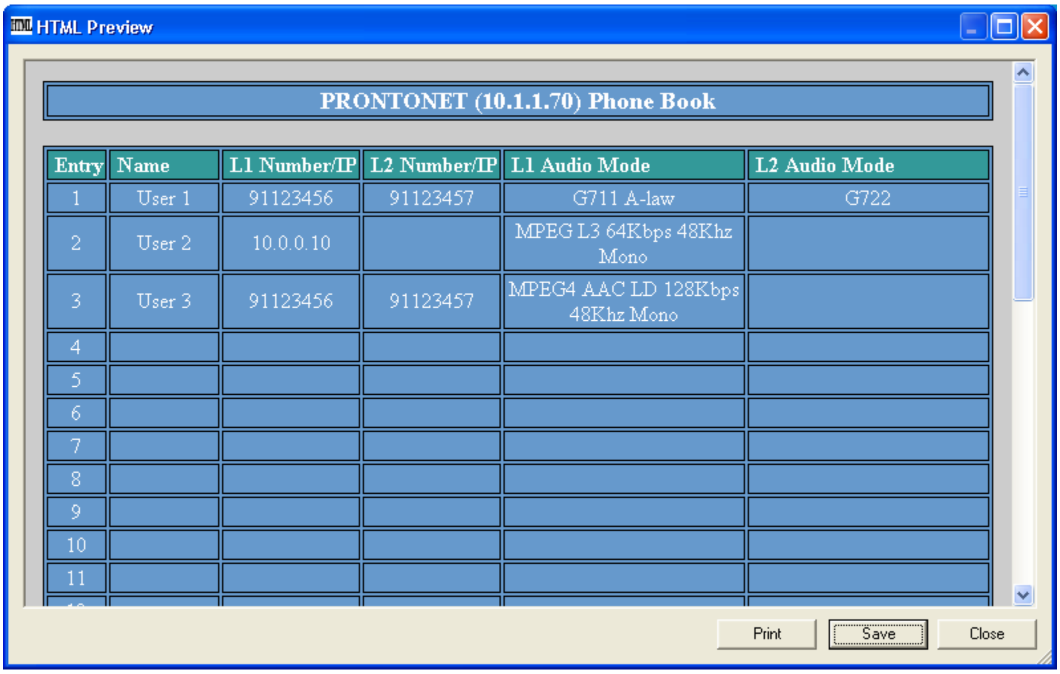


The dial window for line 2 will arrange the information with two different rows for the 'Same contact' (example one) book entry, so the user can select one of both IP addresses.



It is possible to Export / Import the Book in XML format. It is also possible to create an html report allowing the user to preview and print the information.





The audio compression mode is restored to that one prior to the connection. This allows the user to change the compression mode during the connection, or to make calls through the book entries with compression modes different from the former one[13].

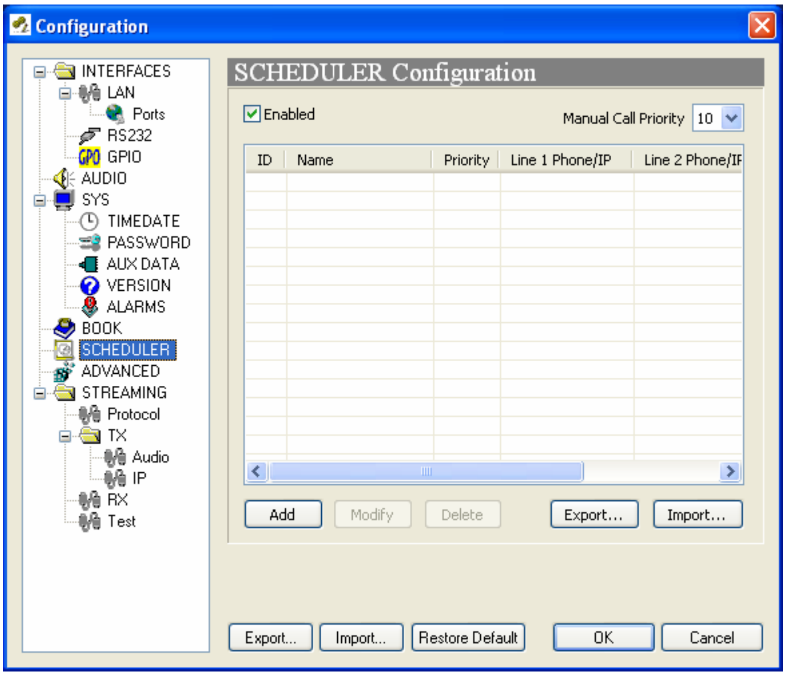
[13] This behaviour was introduced on version 4.8.1.

IV.9 Scheduler

The Scheduler allows the user to program calls to be made automatically. In addition, the duration and the frequency for each call can be defined.

IV.9.1 Configuration

To access the Scheduler configuration window, the user has to click on 'Config' on the web page, and then select "Scheduler" in the general configuration window.



IV.9.2 How to enable/disable the Scheduler

The Scheduler dialog has a checkbox which allows the activation/deactivation of the Scheduler. When the Scheduler system is disabled, none of the scheduled calls will be made. In addition, each scheduled call can be enabled/disabled individually.

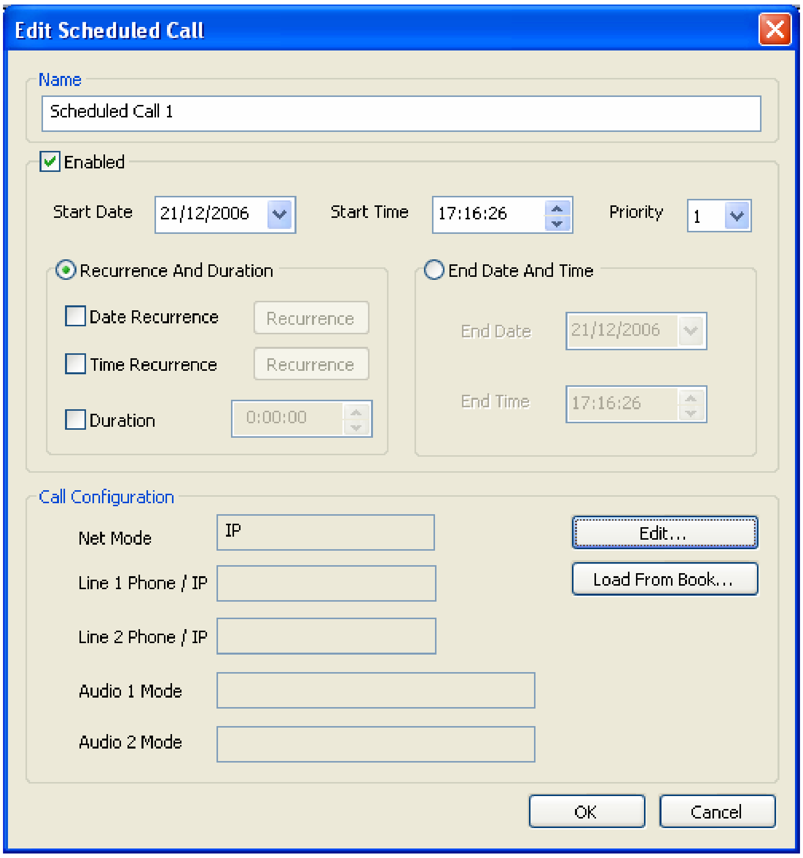
IV.9.3 Automatic and manual call

It is possible to define the precedence between manual and automatic calls. Thus, the user can decide if one automatic call will take precedence over a manual one. The call priority can be set to a number between 1 and 10 (highest priority). The Scheduler will only establish a programmed call when this call has the same or higher priority than the current one. The default priority for automatic calls is 1, and the default one for manual calls is 10. Manuals calls are those made by the user. It is possible to modify the default priority for the

manual calls from the Scheduler dialog, by changing the value of the field 'Manual call priority'.

IV.9.4 Programming the scheduler

To add a new automatic call, click on the 'Add' button in the Scheduler configuration window, or double-click in an empty row. The following window will be displayed:



This dialog permits the user to configure all the parameters related to an automatic call.

- **Name**

This field allows the user to enter a brief description to identify this call. This name will appear in the call list of the Scheduler. By default, the name will be "Scheduled Call" followed by a number.

- **Enable**

Each scheduled call can be enabled/disabled individually. In addition, the Scheduler dialog has a checkbox which allows the

activation/deactivation of the Scheduler. When the Scheduler system is disabled, none of the scheduled calls will be made.

- **Start**

The "Start Date" and "Start Time" fields permit the user to configure the date and time when the scheduled call or pattern will be enabled.

- **Priority**

By clicking on "Priority" the user can change the priority value assigned to the current entry.
It is possible to define the precedence between manual and automatic calls. Thus, the user can decide if one automatic call will take precedence over a manual one. The call priority can be set to a number between 1 and 10 (highest priority). The Scheduler will only establish a programmed call when this call has the same or higher priority than the current one. The default priority for automatic calls is 1, and the default one for manual calls is 10. Manuals calls are those made by the user. It is possible to modify the default priority for the manual calls from the Scheduler dialog, by changing the value of the field 'Manual call priority'.

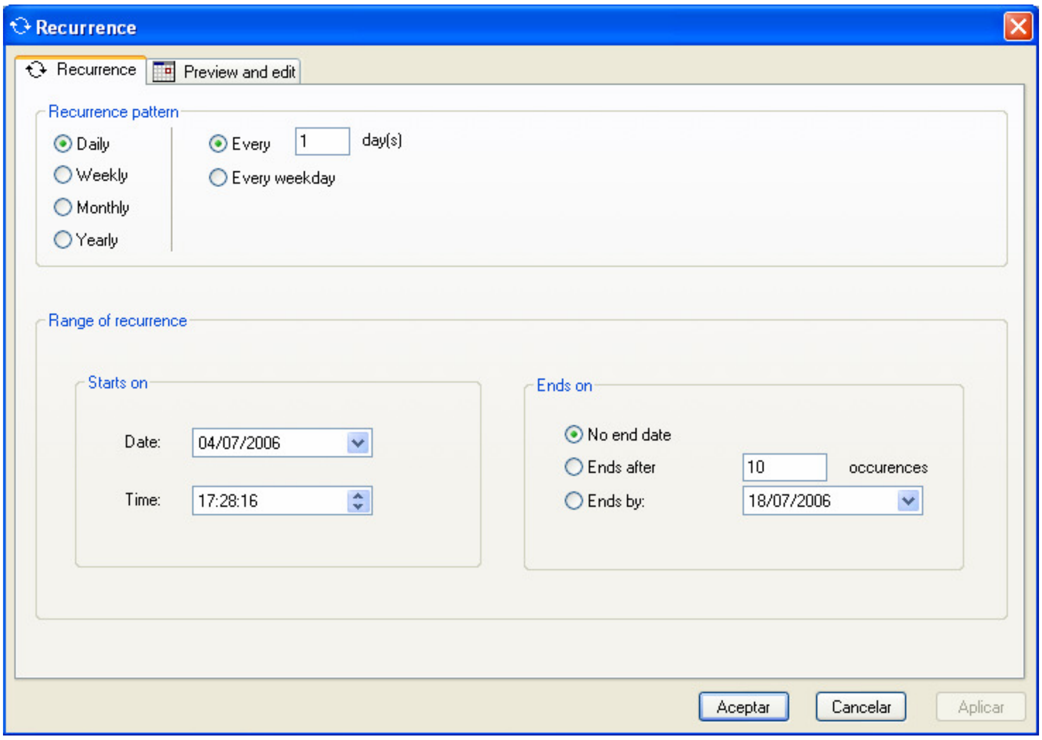
- **End**

There are two methods to define how long the call will last: "Recurrence and duration" and "End date and Time".

- **Recurrence and duration**

This option allows the programming of call repetitions based on time or date: Time recurrence and Date recurrence.

To configure the unit for "Date recurrence" mark the corresponding checkbox in the Scheduler configuration window and click on "Recurrence". The following window will be displayed:



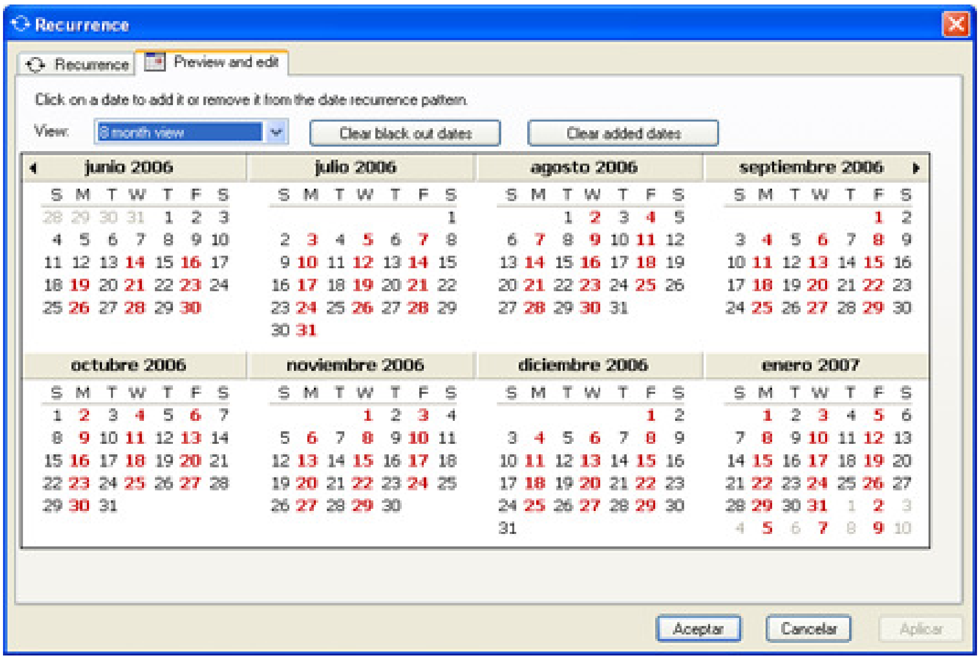
This window is arranged in two different sections:

- "Recurrence pattern": This part allows the user to configure the dates when the call
  will be made. It is possible to program daily, weekly, monthly and yearly patterns.

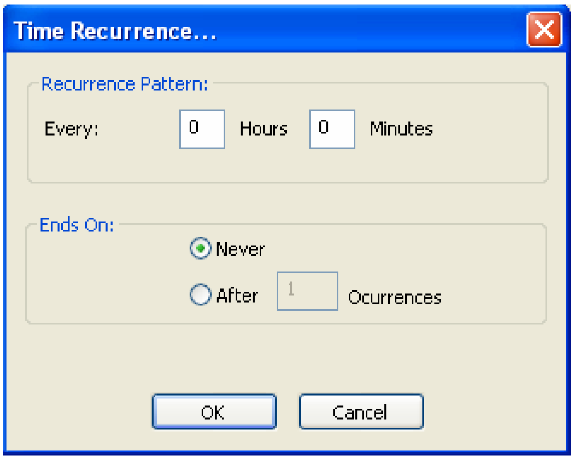
- "Range of recurrence": This section is divided in two different parts:

  - "Starts on": This option establish the start date and time for the configured pattern.
  - "Ends on": It is possible to establish the end of the configured pattern in three different ways: Without end date, after a certain number of repetitions, or in an specific date.

By clicking on "Preview and edit" tab, the user will obtain a calendar wherein those days when this pattern is active will be coloured on red.



To configure the unit for “Time recurrence” mark the corresponding checkbox in the Scheduler configuration window and click on “Recurrence”. The following window will be displayed:



This window is arranged in two different sections:

- “Recurrence pattern”: This section permits the user to configure the period of time for this call to be repeated.
- “Ends on”: From here, it is possible to establish the end of this recurrence. It can be never or after an specific number of occurrences.

- **End date and time**

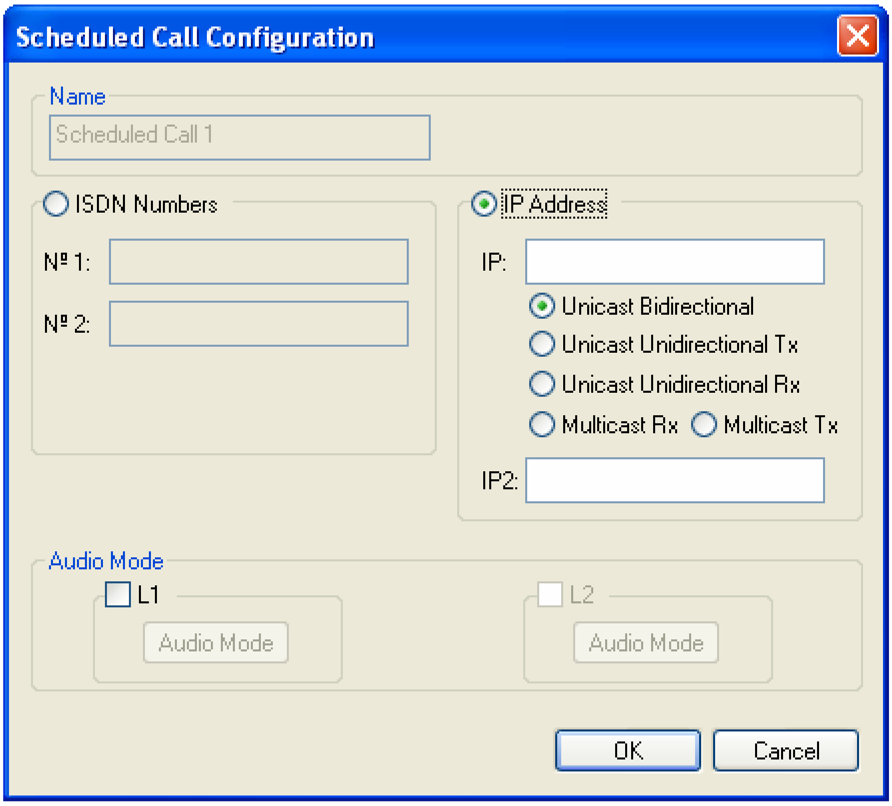
  This option allows the user to establish the duration of a programmed call by defining the end time and date. The call will be established in the date and time defined by the **"**Start Date**"** and **"**Start Time**"** fields.

IV.9.5 Scheduled call configuration

It is necessary to configure the type of network, the numbers and the compression mode that will be used for call establishment. To do that, the user can configure these parameters manually, or automatically, by getting them from one of the entries of the phone book.

IV.9.5.1. Manual configuration

To configure these parameters manually, clicK on "Edit…". The following dialog will be displayed:

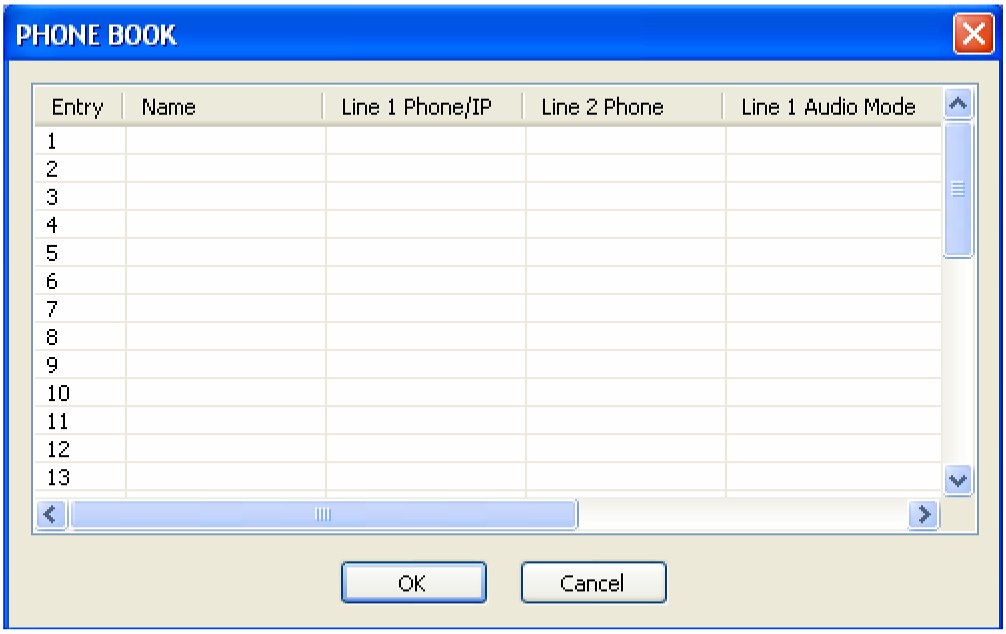


From this window the user can enter all the parameters involved in a call establishment.

IV.9.5.2. Automatic configuration from the phone book

To configure these parameters automatically, click on "Load From Book…" A window with all the entries of the phone book will appear. From this window

the user can select any of the existing entries by clicking on any of them and on the OK button. The selected entry will be read and its configuration will be stored for the current pattern.



IV.9.6 Modifying existing scheduled calls

To modify a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Modify", the user will be able to edit all the parameters related to the select pattern.

IV.9.7 Deleting existing scheduled calls

To delete a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Delete", the select pattern will be deleted.

IV.9.8 Copying and pasting scheduled calls

From the Scheduler configuration window, the user can copy and paste previously configured scheduled calls.

To copy a pattern, select one from the list and right-click on it and select the option "Copy Call(s)".

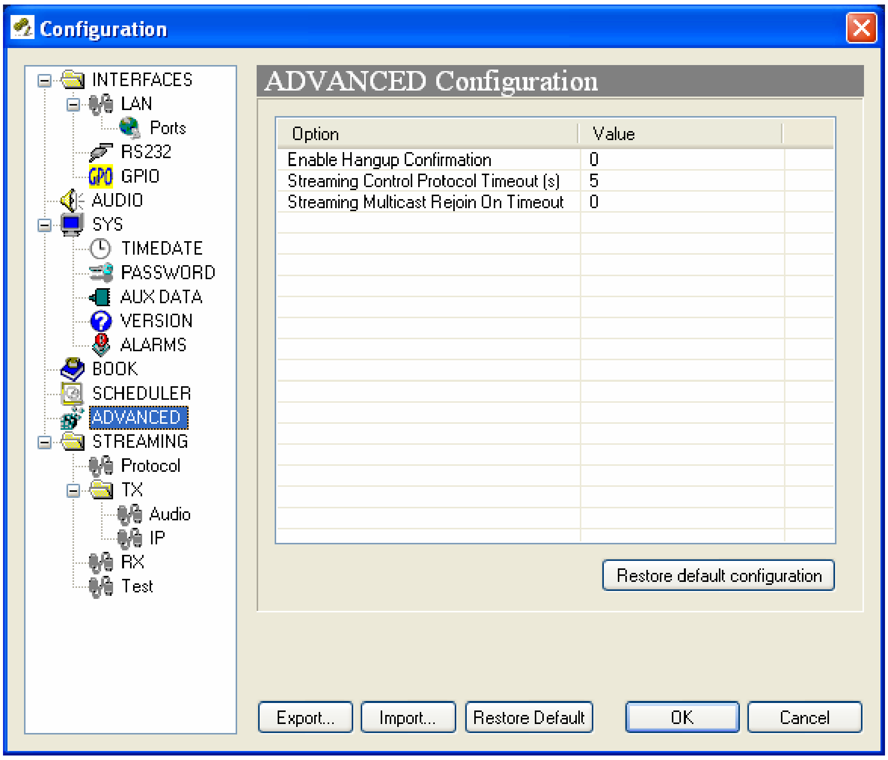
To paste a pattern, select one from the list and right-click on it and select the option "Paste Call(s)".

IV.9.9 Monitoring scheduled calls

When one call related to one of the patterns defined in the scheduler of the unit is established, the message "SCHEDULED CALL IN PROGRESS" will be displayed on the web page for the duration of the call.

IV.10 Advanced

Streaming Multicast Rejoin On Timeout: IGMP protocol functionality.
Enable Hangup Confirmation: If the value is "1", the application will ask confirmation before disconeccting the line.
Streaming Control Protocol Timeout: When the link is down for the specified number of seconds, the audio connection is finished automatically by the unit. This avoid situations where the connection is lost, and the unit still shows 'connected' on the screen.



IV.11 Streaming

Streaming parameters objectives:

- To allow adjustment of the various transmission and reception parameters to optimize the unit to provide the best quality real-time audio streaming for a particular network connection.

- Provide a 'test' tool to check in real time the bandwidth, delay and jitter of the IP connection between two Prodys IP codecs (Nereus, ProntoNet, PortaNet…).

These tools are grouped in the following manner:

1.- Protocol
2.- Transmission parameters.
3.- Reception parameters.
4.- Test tool.

IV.11.1 Protocol

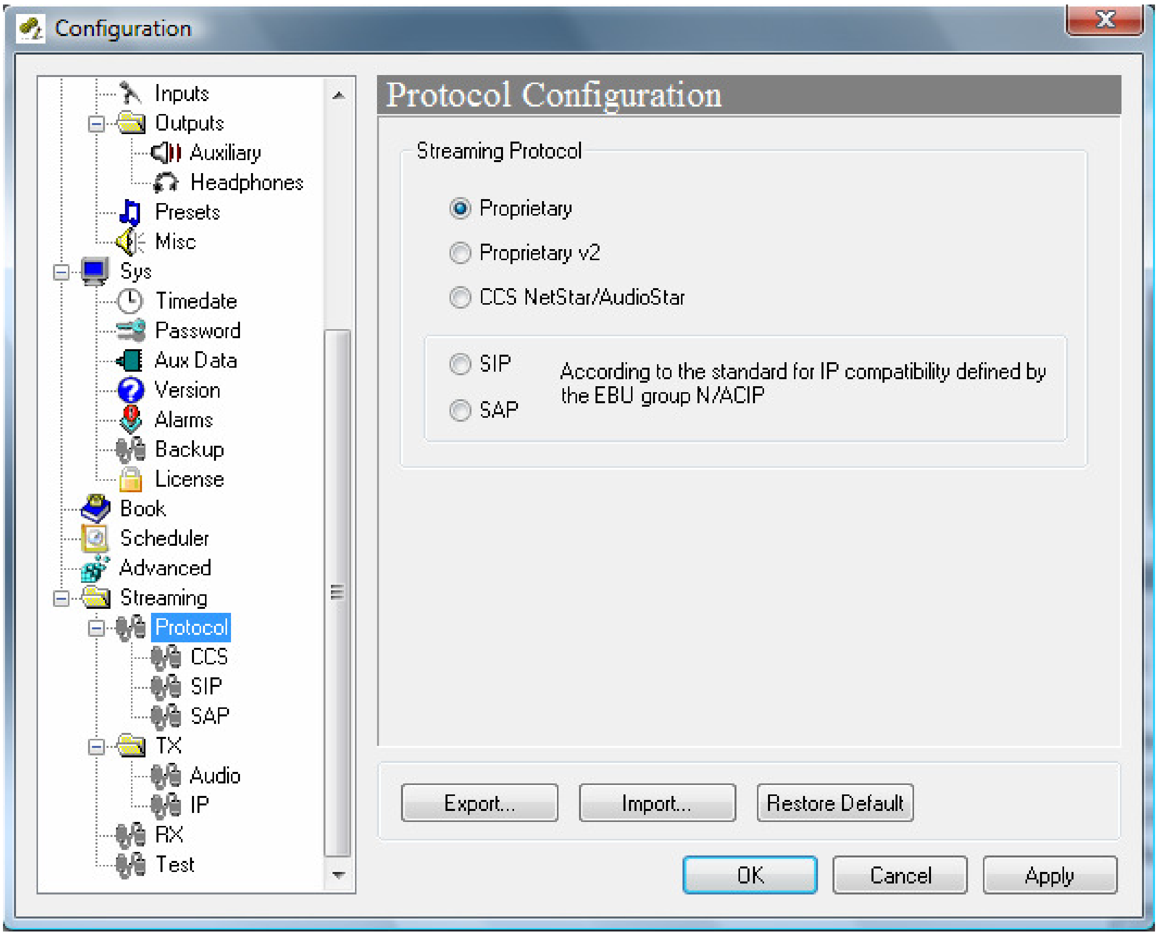
Streaming Protocol: This option allows the user to configure the streaming protocol to use for transmission over IP. There are five choices;

- Proprietary: Prodys proprietary IP streaming protocol.
- Proprietary v2: Prodys proprietary IP streaming protocol version 2.
- SIP[14]: SIP is a signaling protocol for creating, modifying, and terminating sessions with one or more participants. For more information about this protocol and compatibility with other units please contact sales@prodys.net.
- SAP[15]: SAP is a protocol for broadcasting multicast session information.
- CCS NetStart/AudioStar.

The Streaming protocol must be the same at both end of the connection.

[14] SIP protocol is available from version 4.7.1.

[15] SAP protocol is available from version 4.9.0.


Configuration
Inputs
Outputs
Auxiliary
Headphones
Presets
Misc
Sys
Timedate
Password
Aux Data
Version
Alarms
Backup
License
Book
Scheduler
Advanced
Streaming
Protocol
CCS
SIP
SAP
TX
Audio
IP
RX
Test
Protocol Configuration
Streaming Protocol
Proprietary
Proprietary v2
CCS NetStar/AudioStar
SIP
SAP
According to the standard for IP compatibility defined by the EBU group N/ACIP
Export...
Import...
Restore Default
OK
Cancel
Apply

IV.11.2 Audio -Tx

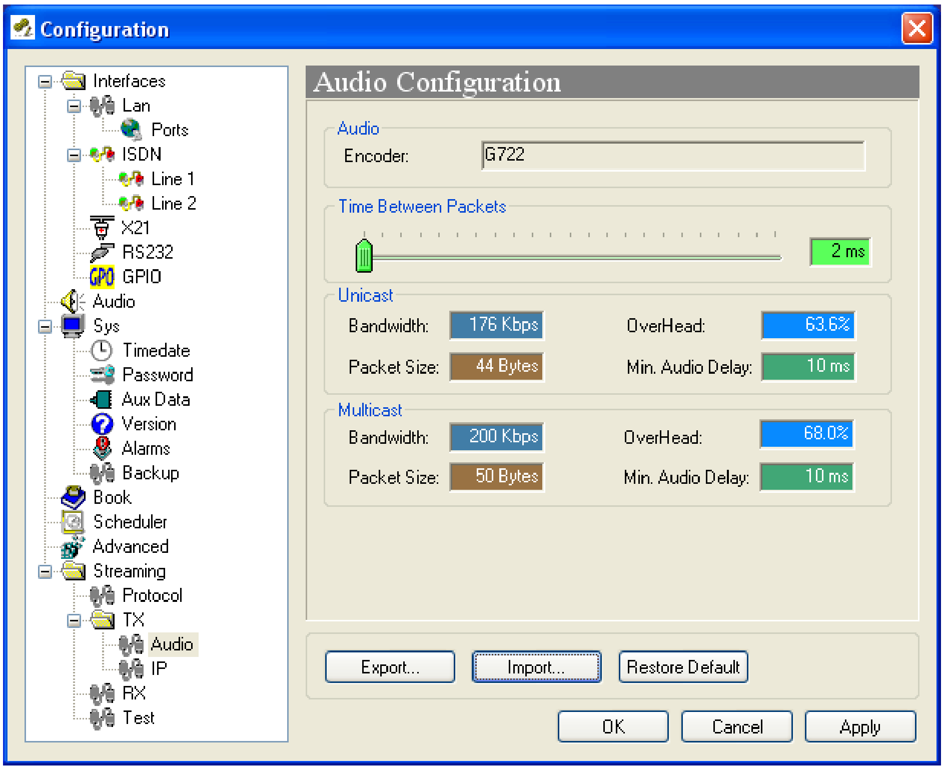
Audio: This option allows the user to configure the audio transmission parameter (TBP: Time between packets) and to know in advance, the required bandwidth and the delay of the connection.

IP: This option allows the user to configure some IP header fields to carry out QoS.

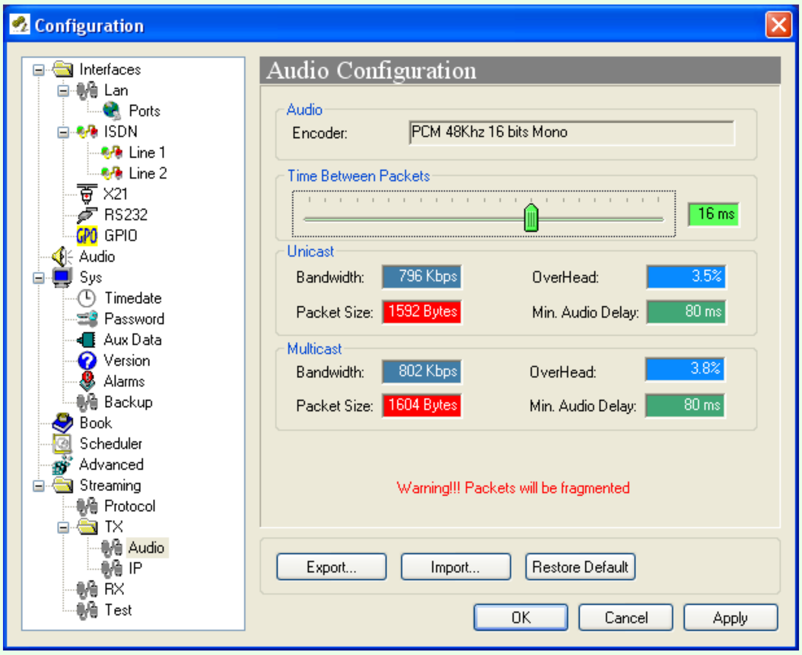
IV.11.2.1. Audio

The 'Time Between Packets' (TBP) parameter is directly related with the packet size and therefore with the occupied bandwidth (overhead) and delay. Therefore, the appropriate value for this parameter is a trade off between delay and bandwidth.

In order to achieve the minimum delay between encoder and decoder it is necessary that the transmission time between packets is minimum as well. However, the greater the TBP, the less bandwidth is required (less overhead), but the more delay there will be in the connection. In other words, the smaller the TBP, the smaller the packet size, and the greater the overhead, due to the IP headers in the packets. With this tool, the user can know beforehand the required bandwidth, overhead, packet size and delay for the current compression mode and TBP value.



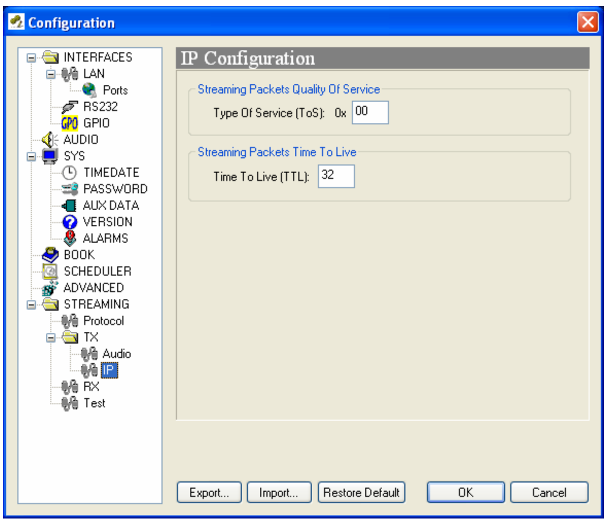
When the packet size needs to be fragmented keep in mind that the overhead changes for that reason. When it happens it is indicated as follows:



IV.11.2.2. IP

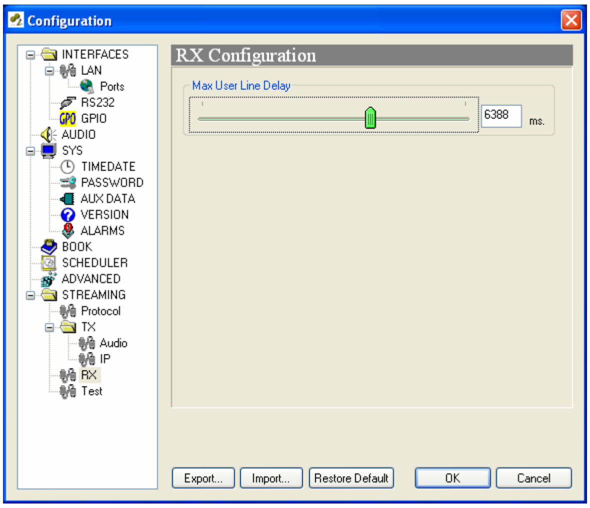
This option allows the user to configure some IP header fields to carry out QoS.

- Streaming Packets Quality of Service: This option allows the user to configure the value in the ToS field in the IP header. This field can be used to carry out QoS in the network.
- Streaming packets Time To Live: Configure the Time To Live parameter.



IV.11.3 Rx

A decisive factor in real time audio streaming is the 'jitter', or delay variation. The 'jitter' is the difference between the maximum and minimum delay. If the jitter value is 0 it means that the delay is constant and it is not necessary to adjust the jitter correction buffer in the reception side. If the delay is not constant, it is necessary to adjust the buffer size in order to guaranty not audio drops, even when the delay reaches the maximum value. To deal with the jitter in the connection, Nereus provides a tool which allows the user to modify the size of the reception buffer, and so, to compensate the jitter. The maximum value for this buffer is 10 sc[16].



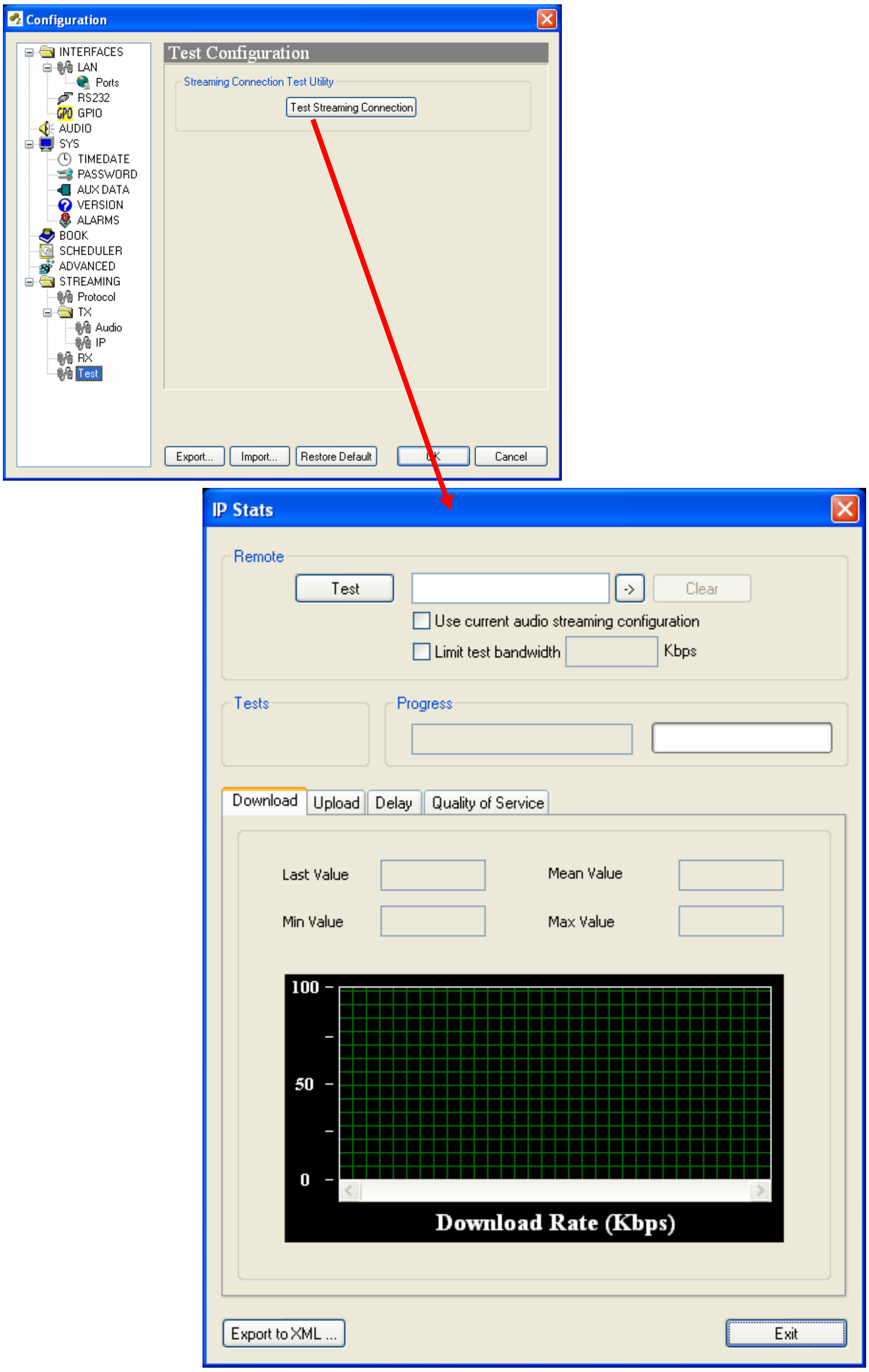
This value can be changed during the audio connection[17].

[16] Before version 4.7.1, the maximum value for this buffer was 500msc.

[17] This option is available from version 4.8.1 onwards.

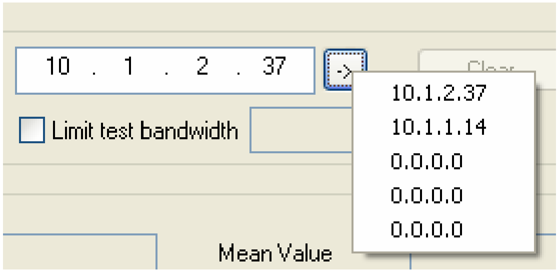
IV.11.4 Test

This tool allows the user to check, in real time, the bandwidth, delay, jitter and packet loss in an IP link between two Prodys IP codecs (Nereus, ProntoNet, PortaNet…). This information will be used to adjust the streaming parameters in order to achieve the best quality in the audio streaming connection.

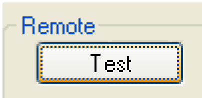


This tool is only available when lines are disconnected.

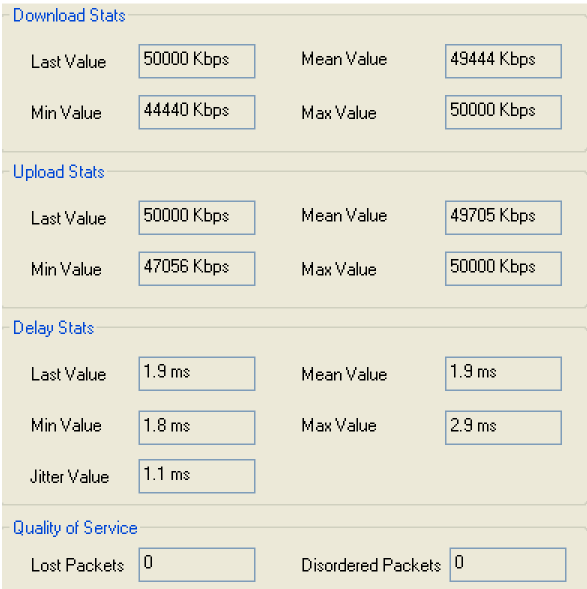
How to run the tests:
1.- First of all the user must enter the IP address of the remote Prodys IP codec. This dialog stores the last used IP addresses.



2.- To run the test just press the test button:

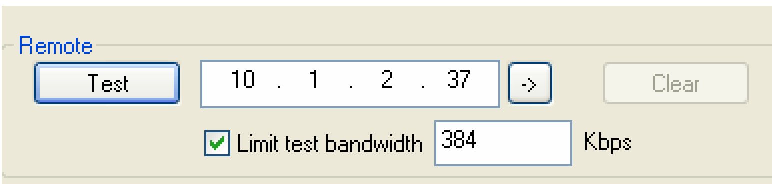


3.- The dialog will show the values:

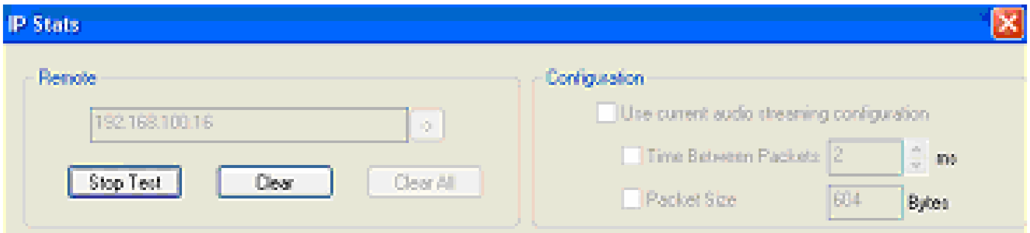


4.- Press "STOP" to stop the test or "CLEAR" to reset the readings.

5.- The user can limit the bandwidth used by the test tool:

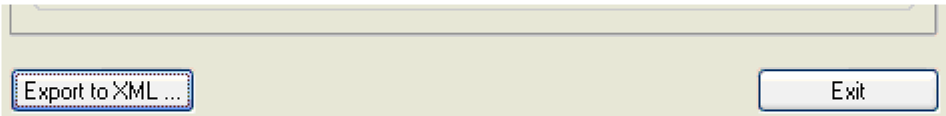


By default this test will attempt to send as much IP traffic as possible. If you are using this test on a "live" network then it is desirable to limit the test bandwidth to that which will actually be used. Thus, it is also possible to simulate the same packet rate and bandwidth as that configured for the audio connection[18].



An accurate figure of total bandwidth used (audio data + overhead) can be obtained from the "Audio Configuaration" screen.

Once the test is finished, it is possible to export the results to an XML file[19].

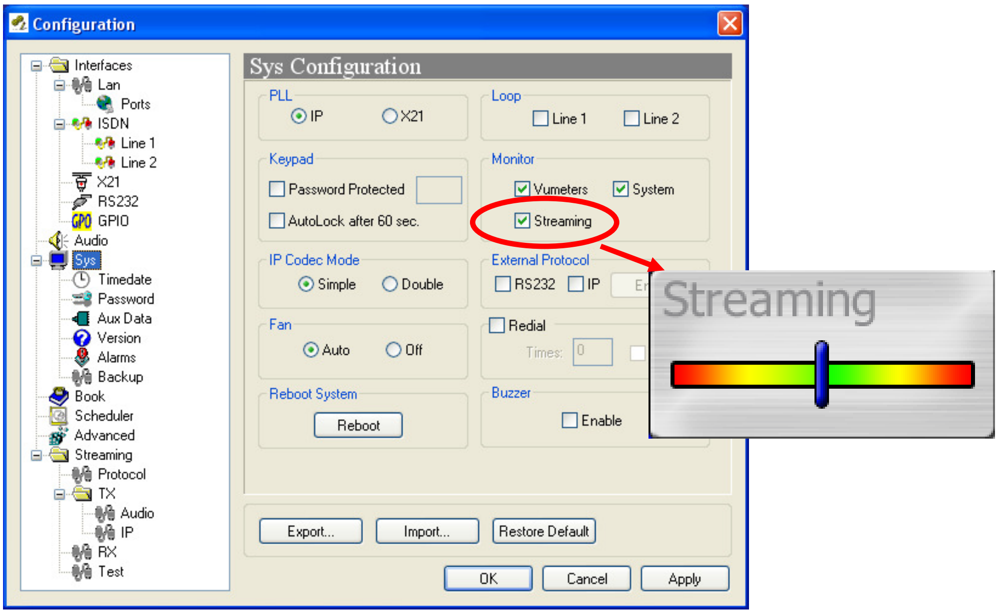


[18] Available from version 5.0.0 onwards.

[19] This option was introduced on version 4.7.1.

IV.11.5 Real Time Monitoring

It is possible to monitor the streaming operation in real time by ticking a checkbox.



That indication represents the instant decoder buffer occupation.

When indication is green or even yellow means the streaming is working fine. Depending on the algorithm, the variation will be quicker or slower but always should be going up and down from yellow to the center (green).

If the bar moves towards the red area, it means that it is getting closer to buffer overflow (red right) or buffer underflow (red left). Of course, if you have buffer overflow or underflow you will have a cut in the audio. If the indication goes to red for an instant, usually doesn't affect the streaming, but of course, if you have a constant red indication you are having problems with the streaming.

Buffer underflow might be caused by lost packets or because the clock of the AD converter at the transmitter side which runs slower than the DA converter at the receiver side.

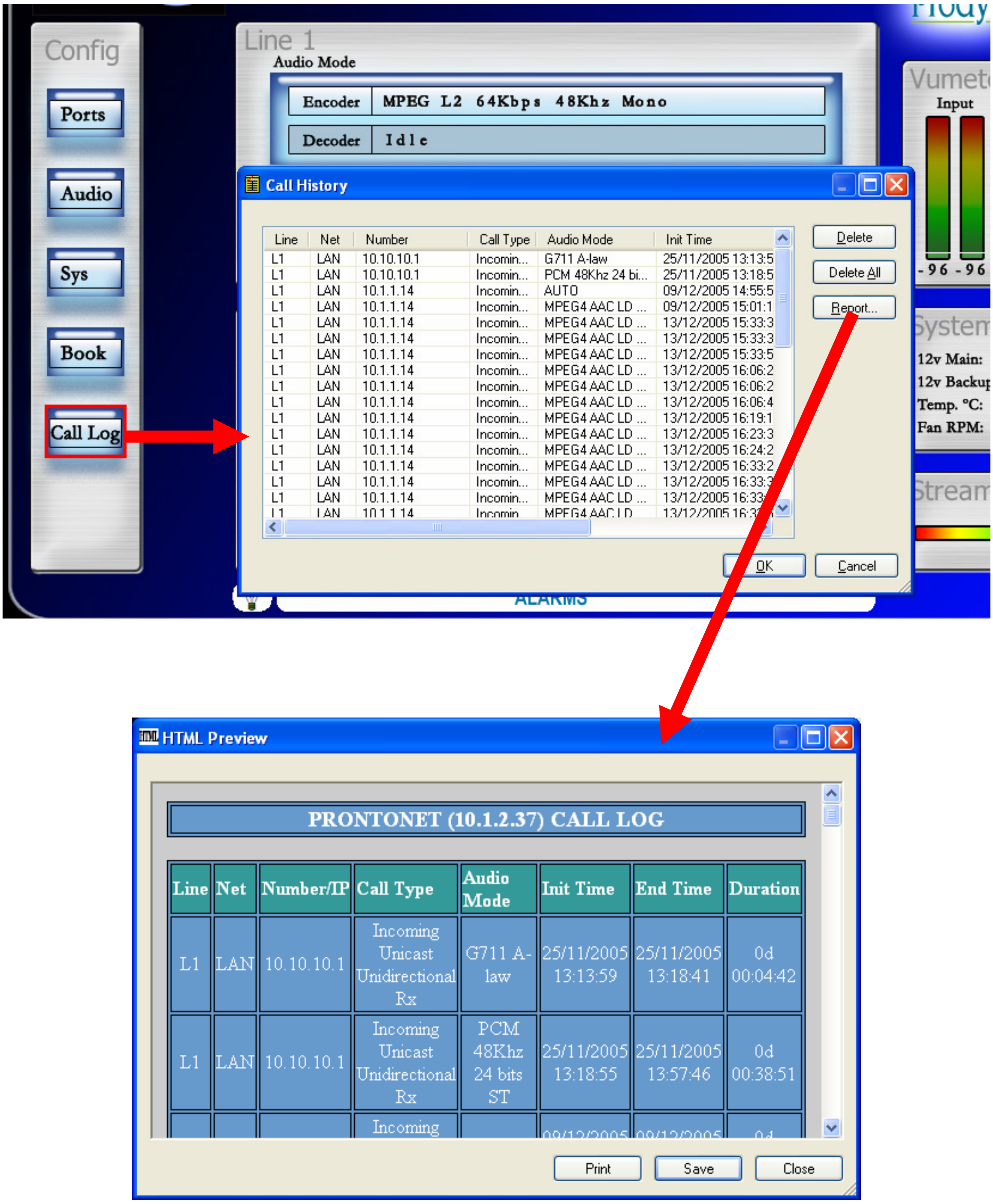
Buffer overflow might be caused by the opposite scheme: The clock at the transmitter side is running quicker than the clock at the receiver side.

Prodys PPL adjustment mechanism is aimed at dealing with this clock jitter and thus to compensate for this variation and guarantee proper audio over IP.

IV.12 Call Log

An history report was included to record the input and output calls according to the following information: telephone/IP, audio modes, date and start/end time, length of each call, etc…

A report of the calls in html can be created allowing preview and print.

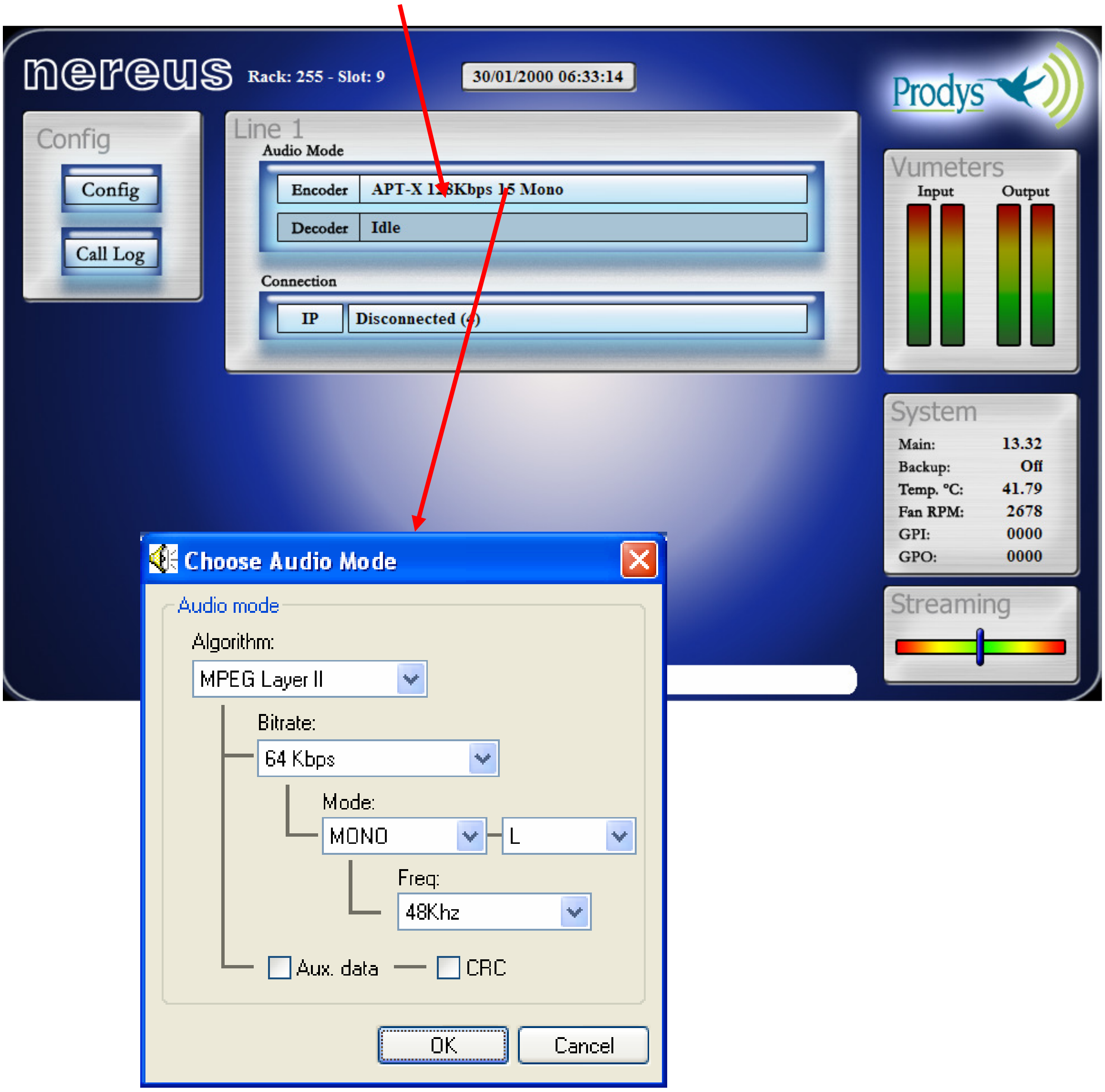


IV.13 Controlling Nereus

From the Control area it is possible to select the communications interface (NET option on the display menu), to configure the encoder or encoders, in case Nereus it is working as Dual Codec, and establish a communication according to the NET selected.

IV.13.1 Configuring the Encoder

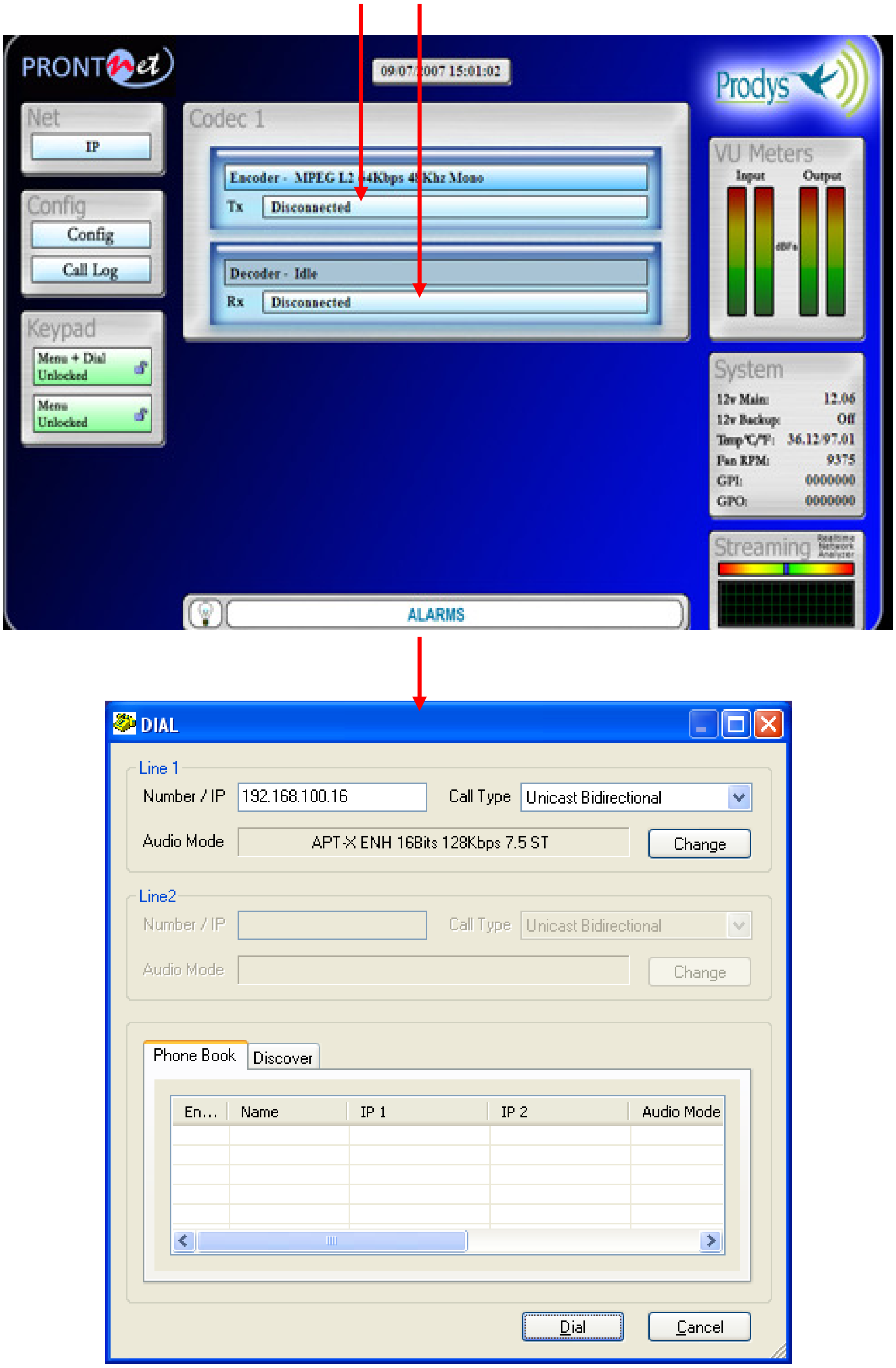
The Audio Mode area shows the Encoder / Decoder current status and also, it allows the user to configure the encoder. By clicking on Encoder area the encoder configuration dialog will be showed:



The encoder configuration parameters depending on the Algorithm.

IV.13.2 Making Calls

By clicking on the right side of the Tx and/or Rx Connection areas[20], the dial menu is shown:



20 The connection bar was split on version 5.2.1 between Rx and Tx for IP connection. If NET = ISDN or X21, the connection bar will be only one, and it will cover Tx and Rx.

Call types:

- Unicast Bidirectional → This is a bidirectional point to point connection, that is, both ends will transmit and receive audio simultaneously. It will be necessary to check the upload and download bandwidth in the link.
- Unicast Unidirectional Tx → This is a unidirectional point to point communication where only the end which makes the call will send audio.
- Unicast Unidirectional Rx → This is a unidirectional point to point communication, where the calling end will be the receiver.
- Multicast Tx → This is point to multipoint communication in which the calling end will join a multicast group as a transmitter.
- Multicast Rx → This is point to multipoint communication in which the calling end will join a multicast group as a receiver.

To make bidirectional or unidirectional Tx calls over IP, click on the Tx connection bar.

To make unidirectional Rx calls over IP, click on the Rx connection bar.

The user can enter the IP address or can select an index from the Phone Book. The encoder can be configured from this dialog as well.

It is possible to discover automatically all Prodys IP codecs connected to the IP network[21]:

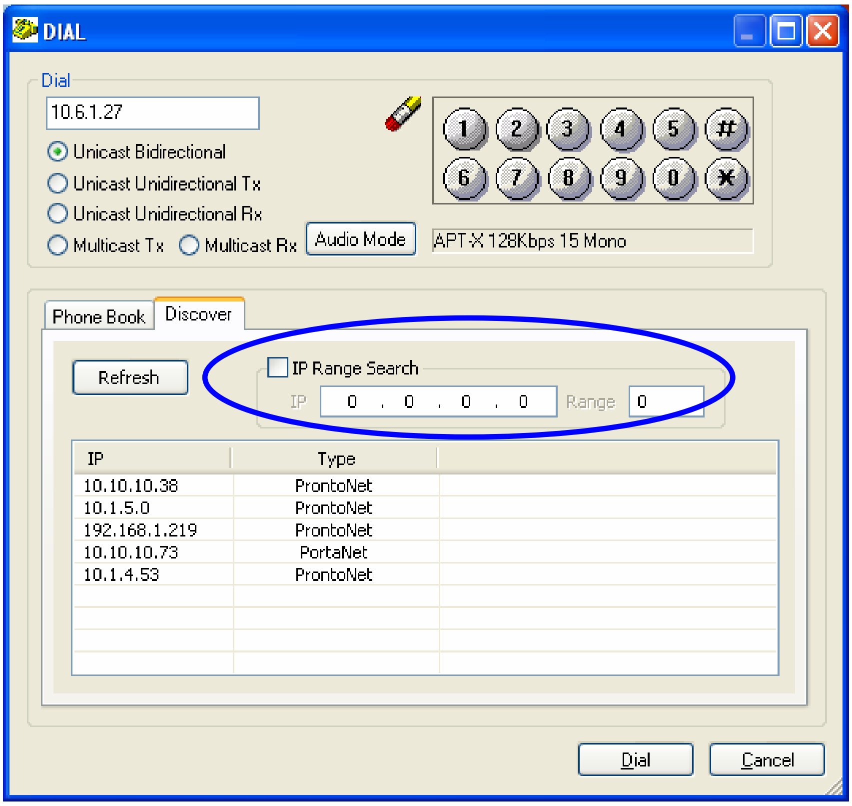
> In order to make easy the dialling operation over IP networks, a new facility was included in the Dial window. This new tool allows the discovering of the Prodys IP units connected to the network.

There are two options to discover the units on the network:

1.- The Nereus sends a broadcast message. This is the default option and it is not necessary to enable anything but **the network must support broadcast traffic.**

2.- When the broadcast traffic is not allowed over the IP network it is still possible to discover the units on the network. The "IP range search" checkbox must be enabled and a range of IP addresses must be introduced for a quicker search.

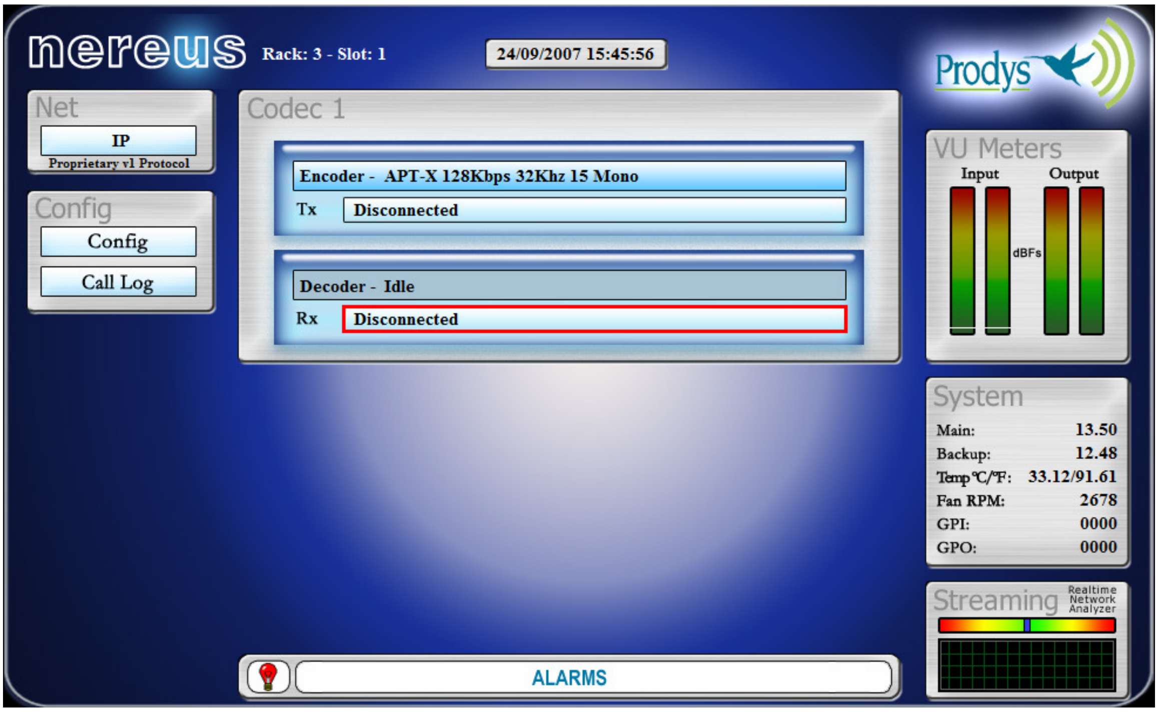
[21] From version 4.7.1 onwards.



IV.14 Monitoring a communication

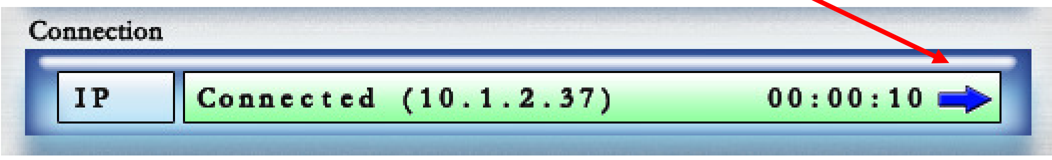
IV.14.1 Line Status

The Line status is shown on the right side of the Connection bar.



These are the different status detected:

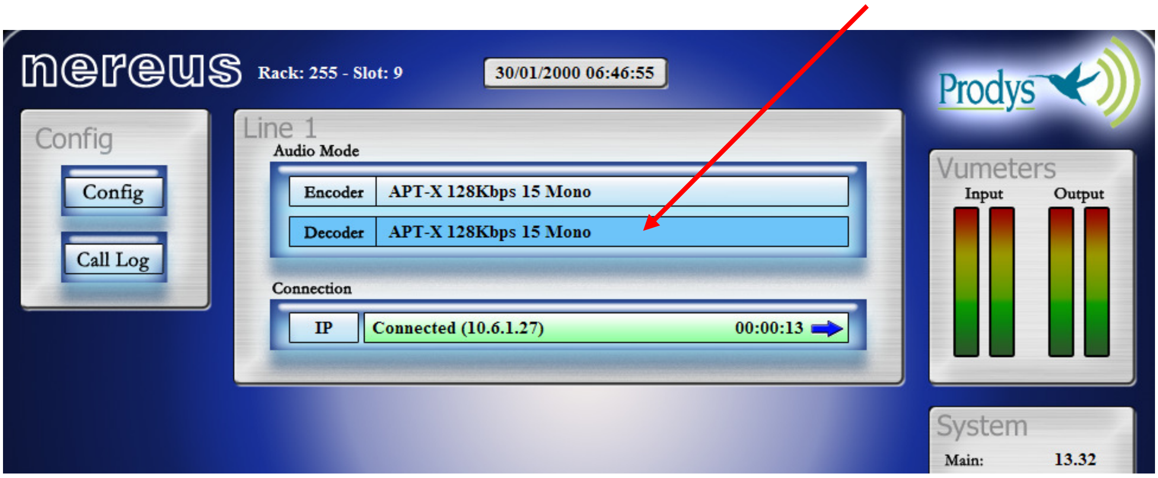
- "No physical Line": The communication line is not physically detected. Most likely the interface is not plugged in. The Display shows "DOWN".
- "Connected":The line is connected. An arrow will indicate if it is an incoming or outgoing call. In the example below, it is an outgoing call.



- "Disconnected": The line is detected physically but no connection is being made. Beside the text appear the disconnection codes.
- "Calling": In the process of making a connection.
- "RING": Receiving a call on the line.

IV.14.2 Decoder Status

The Decoder status is shown on the right side of the Audio Mode area:

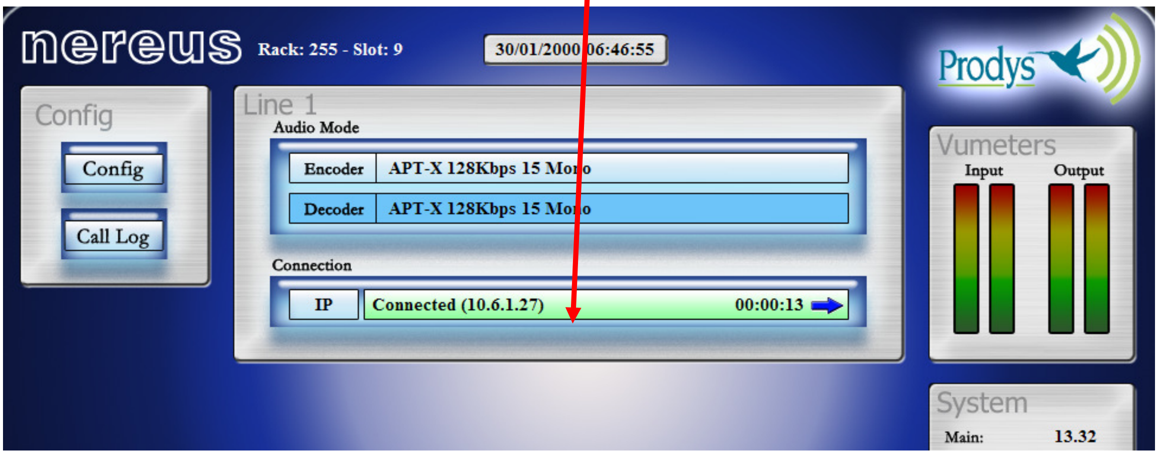


The Decoder status:

- "Searching": The decoder is not synchronized.
- "Framed": The decoder is synchronized. The detected compression mode will be shown on this bar.
- "Idle": The line is disconnected.
- "Not available": Decoder not available.

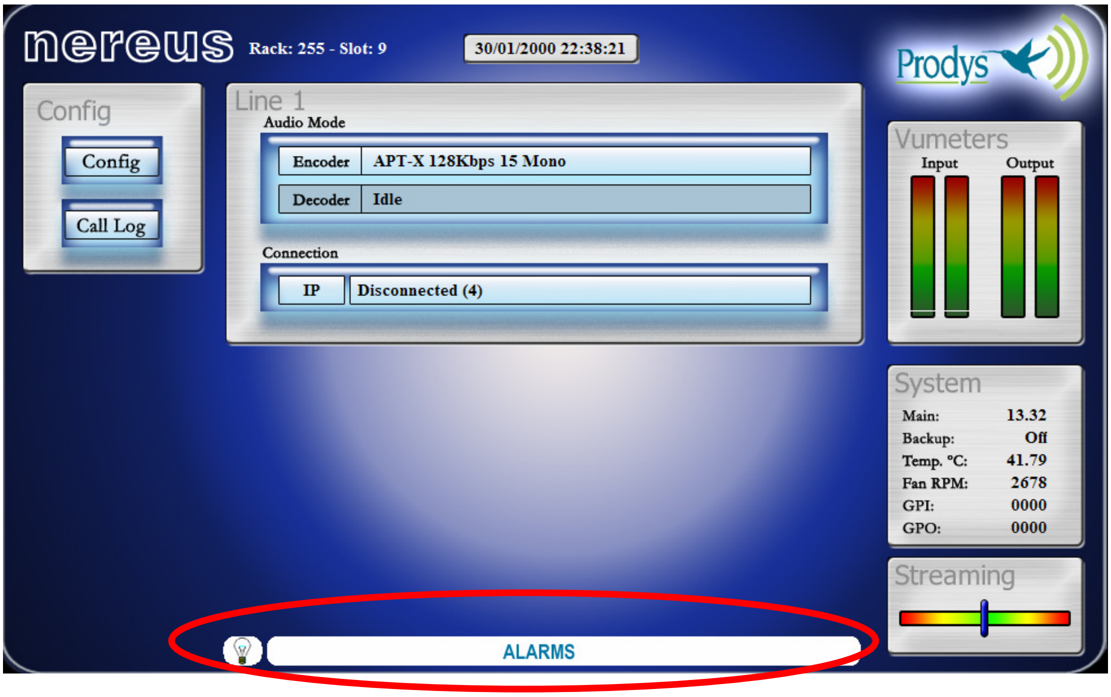
IV.15 Disconnecting the Line

By clicking in the right side of the Connection area.



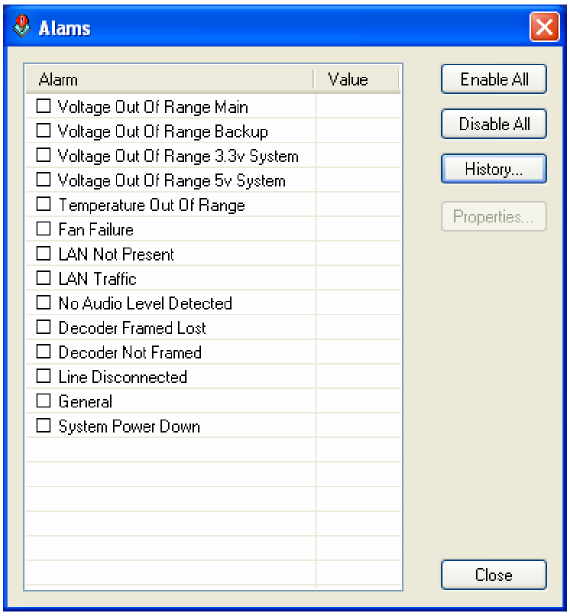
IV.16 Alarms

The alarm window allows the user to select the alarms that the unit will check. The unit will notify the occurrence of each of the selected alarms. It is possible to configure the unit to send SNMP traps or emails to notify alarms information. See chapter IV.7.13 – System Configuration: Alarms.



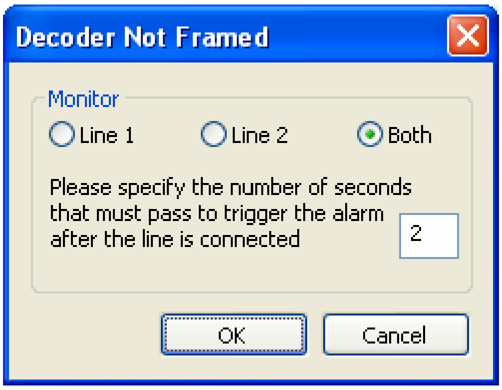
IV.16.1 Selecting Alarms

The unit allows many different alarms. The selection menu is opened by clicking over the Alarm area.



An alarm can be activated for several reasons depending on the selected alarm options. Next the meaning of the alarm options:

- **Voltage Out of Range 12 v Main:** The voltage from the main power supply (AC/DC) is out of range.
- **Voltage Out of Range 12 v Backup:** The voltage from the backup power supply is out of range**.**
- **Voltage Out of Range 3.3 v System:** The voltage from the 3.3 DC/DC converter is out of range**.**
- **Voltage Out of Range 5 v System:** The voltage from the 5 DC/DC converter is out of range**.**
- **Temperature Out of Range:** The temperature is out of range[22].
- **LAN NOT Present:** Not physical level detected in the LAN port.
- **X21 Line Not Present**: Not physical level detected in the X21 port.
- **ISDN NOT Present**: Not physical level detected in the ISDN port.
- **LAN Traffic:** The LAN traffic is higher than the 90% of the capacity of the network.
- **No Audio input level detected**: No audio is present on the input. A threshold and a time period can be defined to activate the alarm.
- **Decoder 1/2 Framed Lost[23]**: The Line is connected but the decoder is not framed, but it was framed before. The alarm will be activated also if the unit is turned off. In this case, the alarm will indicate the date and time when the decoder lost the synchronization, that is, when the unit was turned off. The alarm will be activated until the unit is framed once the line is connected again. If the connection is not possible, the alarm will be marked as finished.
- **Decoder 1/2 Not Framed**: The Line is connected but the decoder is not framed. This new alarm doesn't require that the decoder was framed previously. For this reason, it is necessary to specify a window time to decide when the alarm will be activated once that the NOT FRAMED condition has been detected.



[22] The temperature is above 50ºC.

[23] This alarm was introduced on version 4.7.1.

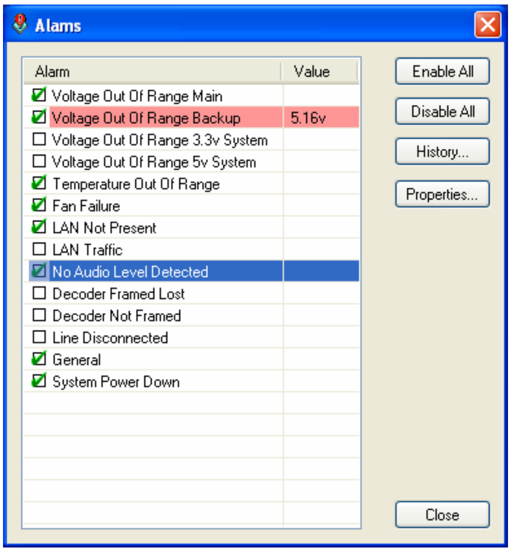
Otherwise, the alarm would come up as soon as the connection is established, given that the Decoder needs some time to detect the incoming compression mode.

- **Line Disconnected**: Line 1/2 has been disconnected unexpectedly, manually disconnected or both.
- **Backup Active**: The ISDN backup line is working.
- **General**: In internal hardware error was detected by the software.
- **System Power Down**[24]: This new alarm allows the user to know if the unit was turn off and the time that it remained in that condition.

IV.16.2 Monitoring Alarms

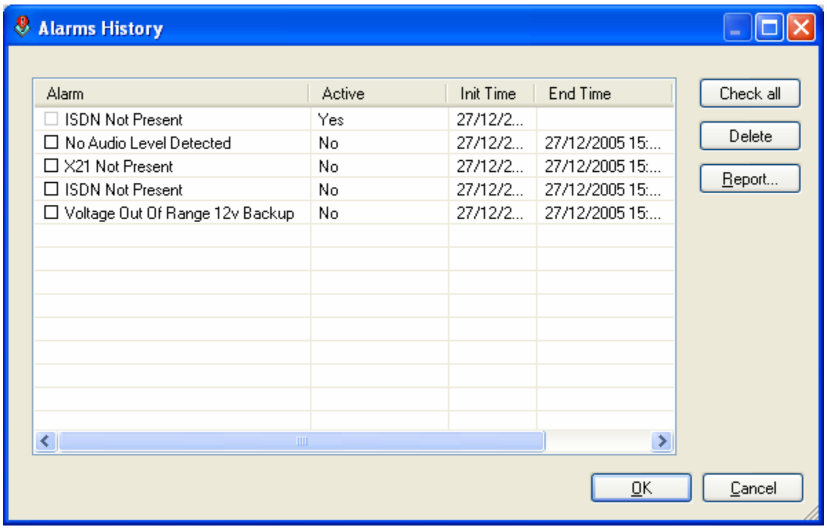
When one of the selected alarms is detected, the alarms window is highlighted:

By clicking over it, the information will be showed:

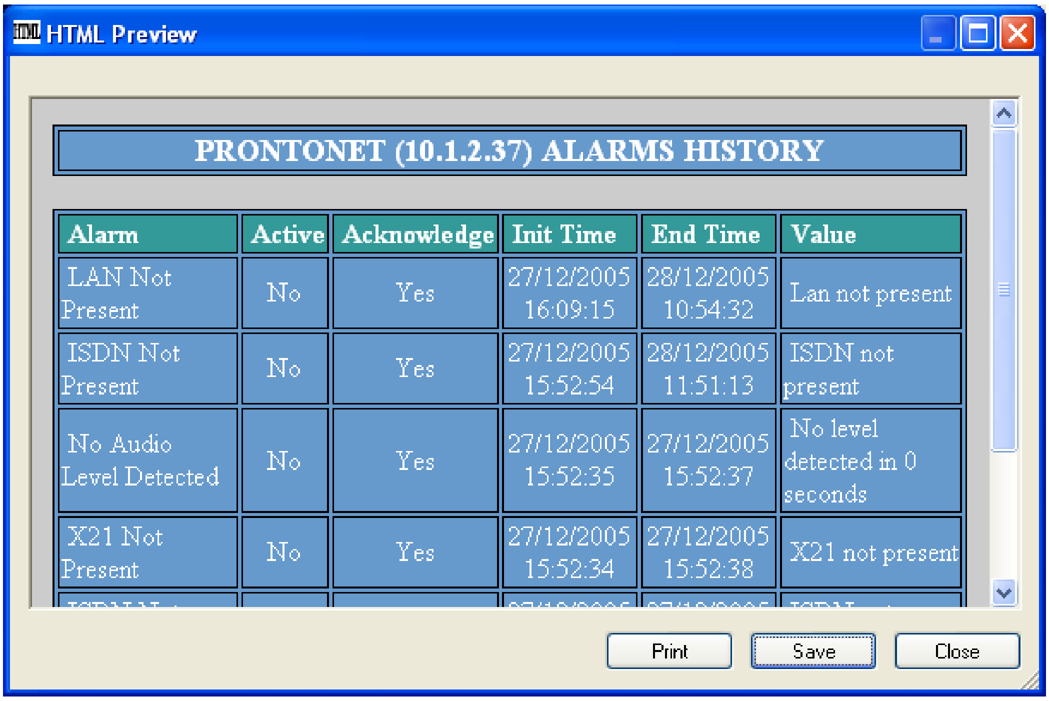


The alarms are saved on the non volatile memory of Nereus. It means that it is possible to know what happened on the past even if the unit was disconnected of the network. The alarms history can be showed by clicking over the lamp or over "History" option in the alarm selection dialogue.

[24] This option is available from version 4.7.1 onwards.



An historical report of alarms in html allowing preview and print is available.



How does Nereus work?

Given the different nature of the three communication standards that the Nereus supports, the unit has to adopt different configurations while adapting to the characteristics of each network. What's more, in certain situations, Nereus is capable of dealing with two independent communications at the same time. This can be a cause of confusion for the user, who may be expecting a certain behaviour or action from the unit but observes another. This chapter is therefore a practical guide to help in understanding just how the Nereus unit works under different configurations, especially the more unusual ones.

V.1 Selecting the communications interface

Firstly, when we refer to the 'communications interface' we are always referring to the communications port that is used for the transmission of 'audio data'. The NET option in the main menu is for setting this up. There are three settings available with the Nereus:

a. Nereus operating as "IP CODEC": The Nereus will be configured to establish communications via the Ethernet port using the IP protocol.
b. Nereus operating as "ISDN CODEC": The Nereus will be configured to establish communications via a basic access ISDN connection. The Nereus has a multi-protocol ISDN terminal socket ready for connection to a basic rate 2B+D interface.
c. Nereus operating as "X21 CODEC": The Nereus can be configured to establish communications via dedicated digital lines at 64, 128, 192, 256, 384 and 576 Kbps.

The Ethernet interface is always available for remote control of the unit, even if it is configured to operate as an ISDN or X21 codec.

You should first select the network type - i.e. the communications interface - to be used for audio data transmission and reception. The menu will only show the configuration options relevant to the network type selected. Of course, there are a number of general parameters that will be the same across all network types and so will not be affected when the network type is changed.

V.2 Configuration parameters that are dependant on the network type selected

There are two fundamental reasons for which the user must change the Nereus configuration parameters when a new communications interface is selected:

- The available bandwidth over the selected network (IP, ISDN or X21). This places limitations on the bit-rate that can be selected and the amount of compression. As an example: over ISDN it is not possible to send uncompressed audio and the maximum bit-rate for any algorithm is 128Kbps.

- The number of available channels: Number of bidirectional communications which can be achieved. Note that over ISDN and IP, up to two independent mono communications can be achieved.

The following table shows what the differences between the various communications networks regarding Nereus operation:

NET	ALGORITHM	BIT RATES (Kbps)	Nº CHANNELS
IP	PCM	16 bits @ 48 KHz.	2 Mono[25] or 1 Stereo
	G711	64	
	G722	64	
	MPEG1,2 LII	64,128,192,256,384	
	MPEG1,2 LIII	64,128,192,256	
	MPEG AAC LC.LD	64,128,192,256,384	
	MPEG 4 AAC HE	24,32,48,56,64,128	
	aptX™	64,128,192,256,384,576	
ISDN	G711	64	2 Mono or 1 Stereo
	G722	64	
	MPEG1,2 LII	64,128	
	MPEG1,2 LIII	64,128	
	MPEG 2,4 AAC LC	64,128	
	MPEG 4 AAC LD,HE	64,128	
	aptX™	64	
X21	G722	64	1 Mono or 1 Stereo
	MPEG1,2 LII	64,128,192,256,384	
	MPEG1,2 LIII	64,128,192,256	
	MPEG AAC LC,LD	64,128,192,256,384	
	MPEG 4 AAC HE	24,32,48,56,64,128	
	aptX™	64,128,192,256,384,576	

[25] Over IP, the second MONO communication is fixed to G722.

V.3 Nereus working as a "DUAL CODEC" over ISDN

When we select ISDN as the communications interface, it is possible to establish two totally independent MONO connections for each 64 Kbps B Channel. This means that for each channel or line of communication (Line 1 or Line 2 as seen by the Nereus) it is possible to send an audio signal encoded with any of the available codec algorithms. The menu offers the option to configure Encoder 1 and Encoder 2 separately. However, the user must remember that in this mode it is only possible to work with MONO signals on each channel, given that Nereus has only one stereo audio input and one stereo audio output. To avoid incorrect configuration there are therefore certain restrictions while working with the Nereus as a "DUAL CODEC":

- If Encoder 1 is configured in any mode other than DUAL or STEREO/JSTEREO the Encoder 2 option is disabled and the Nereus will not work as a "DUAL CODEC".

The Encoder 2 is not available also when NET = ISDN and bit rate = 128 Kbps.

- Encoder 2 can only be configured to work in MONO.

Section V.9looks at how the Nereus operates as a ISDN CODEC in more detail, and it will detail other restrictions that need consideration when we configure Encoder 1 in DUAL or STEREO modes, or when it receives calls from other units that are working in any of these modes.

V.4 Nereus working as a "DUAL CODEC" over IP[26]

There are two different operation modes: SIMPLE and DOUBLE mode[27]. In SIMPLE mode, the unit will work as usual, that means, it cannot establish two independent communications via IP. In DOUBLE mode, the unit can establish two independent communications via IP.

In DOUBLE mode over IP, the following restrictions will be applied:

1.- Encoder 1 and Encoder 2 can only be set in MONO.
2.- Encoger 2 (assigned to line 2) can only be set in G722.

This configuration can be set both from the menu and from the Web Page.

[26] Operation mode only available when selecting Prodys Proprietary Protocols over IP.

[27] From version 4.5.0 onwards.

V.5 About how the Decoder works and automatic searching

The Nereus Decoder system doesn't need to be configured as it is totally automatic. However, it is necessary to point out some aspects of how it works as it depends on the format of incoming data. Nereus is able to synchronise automatically to the following algorithms under the following conditions:

Algorithm	**IP Codec (NET = IP)**	**ISDN Codec (NET = ISDN)**	**X21 Codec (NET = X21)**
G711	YES	YES	NO
		The system detects incoming calls such as voice calls and configures the Encoder and Decoder to G711.	Algorithm not available.
G722	YES	YES	YES
		Needs G722 encoded audio since it is synchronised by means of statistical framework. **It does not work if encoder is configured to aptX , AAC HE or J52.**	Needs G722 encoded audio since it is synchronised by means of statistical framework. **It does not work if encoder is configured to aptX, AAC HE or J52.**
MPEG1,2 LII MPEG1,2 LIII MPEG2,4 AAC LC & LD	YES	YES	YES
		Detects all modes excepts if the Encoder is configured to aptX. **It does not work if encoder is configured to aptX, AAC HE or J52.**	Detects all modes excepts if the Encoder is configured to aptX. **It does not work if encoder is configured to aptX, AAC HE or J52.**
MPEG AAC HE & aptX^{TM}	YES	NO	NO
	AptX not with SIP/SAP as IP protocols	It is necessary that the Encoder is also configured to AAC or aptX.	It is necessary that the Encoder is also configured to AAC or aptX.
PCM Linear Audio	YES	NO	NO
		Not Available.	Not Available.

Taking into account the previous table the user must also bear in mind the following:

1. Over X21 or ISDN, it is only possible to set different encoding and decoding algorithms independently, with those algorithms that support automatic synchronising. For example, it is possible to transmit on G722 and receive on MPEG Layer II.

   Although G711 is detected automatically in ISDN mode, the Encoder is also configured automatically because it is not possible to use it in any other combination.

2. If we want to receive an algorithm that does not support automatic synchronisation it is necessary to configure the Encoder to the same algorithm. In the case of apt-X, the compression parameters on the encoder should match those on the decoder.

3. If we want the Encoder to switch to the algorithm detected by the Decoder, the Encoder should simply be set to AUTO mode.

   AUTO does not work with those algorithms that do not support automatic synchronization.

V.6 The Nereus operating as IP codec (Proprietary protocols)

The operation of the Nereus as an IP codec offers three operational modes: UNICAST, MULTICAST and MULTI-UNICAST.

V.6.1 UNICAST communications

The term UNICAST is used in the networking world to refer to the connection to a single destination. Applied to Nereus, this is when a point-to-point connection is created between two units bi-directionally.

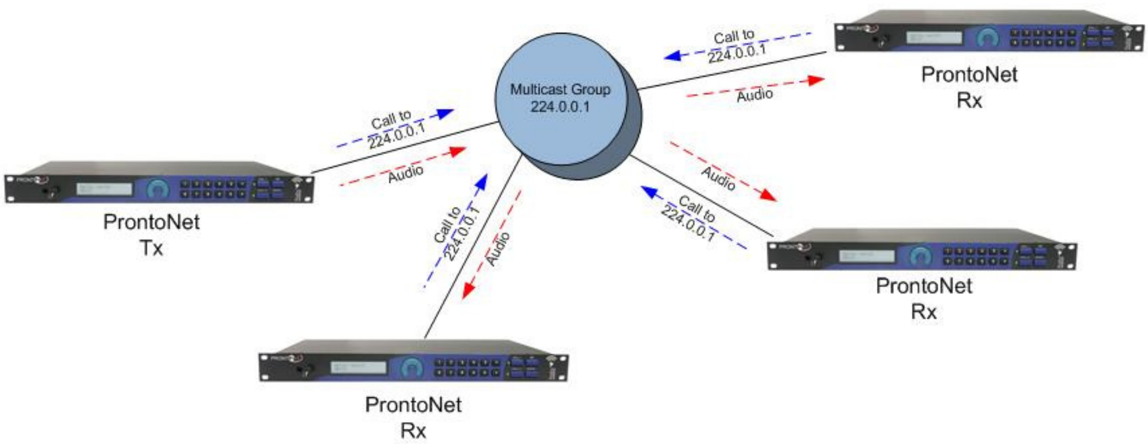
V.6.2 Establishing a UNICAST connection from Nereus

The procedure for establishing a connection is very similar to that of making an ISDN call; we enter the call initiation menu by pressing the CALL 1 key and enter the IP address of the unit we wish to connect to. As we have already mentioned, the audio data connection is bi-directional (as in an ISDN codec) and we need two connections, one in each direction. For this, the Nereus unit that

receives and accepts the call will automatically call back to the originating Nereus and establish a reverse connection.

V.6.3 Establishing a MULTICAST communication from Nereus

With MULTICAST the calls must be made from both ends. Both the sender of the data and all the receivers of the data must call to establish a connection to the multicast group. The multicast operation can be shown in the following diagram:



To initiate audio distribution the role of the transmitter unit and the roles of all the receiver units must be set up for the group. These modes can be selected from the menu **CONF/PORTS/LAN/MULTICAST** where you select **Tx** (transmitter) or **Rx** (receiver). Once this is done, the calls can be established. This can be done in any order, that is, calls can be set up first from the transmitter and then from each of the receivers, or the other way round. The important thing to note is that when the transmitter is connected to the group, it will start the streaming of audio data immediately. If there is no receiver connected, the audio will simply not be received. As soon as receivers are connected to the multicast group they will receive the audio data that is being streamed. Equally, if the receiver is connected to a multicast group where the transmitter is not operating yet, the audio output is muted.

- **Multicast considerations:**
  - Internet Protocol (IP) multicast is a bandwidth-conserving technology that reduces traffic by simultaneously delivering a single

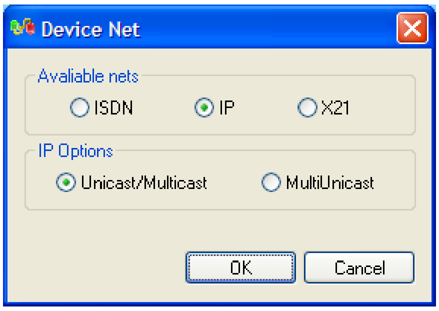
stream of information to thousands of corporate recipients and homes.

- Multicast is based on the concept of a group. An arbitrary group of receivers expresses an interest in receiving a particular data stream. This group does not have any physical or geographical boundaries—the hosts can be located anywhere on the Internet. Hosts that are interested in receiving data flowing to a particular group must join the group using IGMP. All this is done automatically by Nereus when establishing a connection.
- Multicast traffic is rejected when going through the Internet, since most IP servers on the Internet do not currently support the multicasting part of the protocol, except when using VPNs, because VPN's encapsulates IP packets as unicast frames, so routers simply see an ordinary packet.
- All IP multicast group addresses will fall in the range of 224.0.0.0 to 239.255.255.255, but some of them are reserved, that's why the range of addresses from 224.0.1.0 through 238.255.255.255 are called globally scoped addresses.
- There should only ever be one transmitter connected to a MULTICAST group or else audio reception errors will occur.
- For transmitting MULTICAST audio the Prodys proprietary protocol "Prodys eXtended Real Time Protocol (PX-RTP) will be used.
- The Nereus transmitter cannot be in automatic encoding mode.
- To guarantee a constant delay all the units must synchronise their clocks. Each receiver will activate a clock-sync algorithm that adjusts its PLL (Phase-Lock Loop).

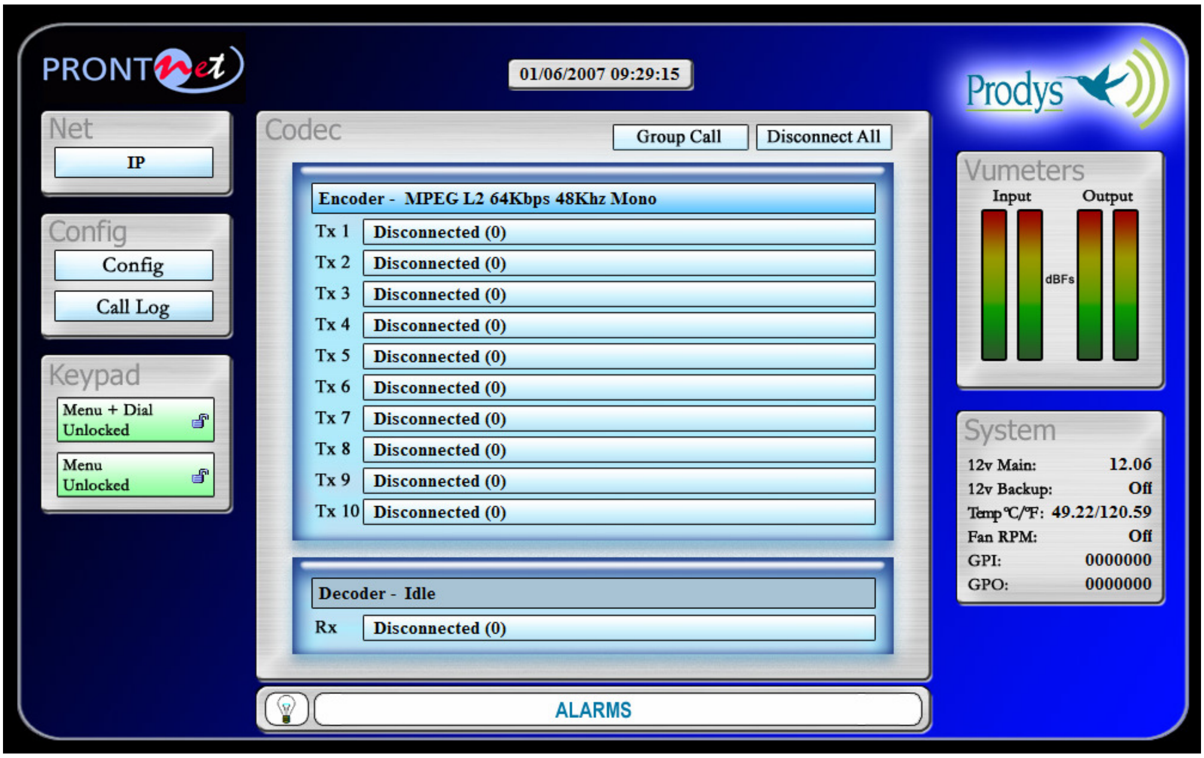
V.6.4 MULTI-UNICAST

MULTI-UNICAST allows the user to establish point to multi-point connections by means of unicast connections. The same audio (one encoder) can be sent up to 10 different locations concurrently. This functionality overcomes the lack of Multicast support on some networks.

To select this new feature, open the new 'NET' menu by clicking on the 'NET' button at the top on the left in the web page.



The main window will show all the available connections:



Depending on the compression mode, up to 10 Tx connections will be available:

- MPEG L2, L3 and AAC: Up to 10 Tx + 1 Rx.
- PCM, G711, G722, APTX: Up to 3 Tx + 1 Rx.

As can be seen, there is an independent control bar for each connection. Each of the connection bars will show the line status in real time. The procedure to establish a call is the same as that for Unicast connections.

As for Unicast connections, the Tx and Rx parts of the connection are divided so that it is possible to receive from one end at the same time the unit is sending audio on one or more Tx connections.

In addition, it is possible to make or hang up several connections at the same time by using the 'Group Call' and 'Disconnect All' buttons respectively.

This operation mode is not compatible with the IP 'DOUBLE MODE'.

V.6.5 Prodys Proprietary set of protocols

Prodys has developed this proprietary set of protocols to carry out IP streaming connections, due to the lack of a standard in this regard:

- Prodys Real Time Control Protocol (P-RTCP): This is a protocol based on TCP that allows for the establishment and termination of a connection as well as for the negotiation of the codec mode (automatic audio synchronisation in all modes).
- Prodys Real Time Protocol (P-RTP): This is a protocol based on UDP used for the transmission of audio.
- Prodys eXtended Real Time Protocol (PX-RTP). This is a protocol based on UDP used for the transmission of multicast audio.
- Prodys Upgrading/Identifying Protocol. This protocol is based on UDP and used to identify/upgrade the units.
- Prodys External Protocol (P-XP). This protocol is based on TCP and can be used for controlling the units from an application other than the web page or ProdysControl.
- Prodys Auxiliary Data Protocol (P-AUXP). This protocol is based on UDP and used for transmitting/receiving auxiliary data. Prodys U-bit Protocol (P-UbP). This protocol is based on UDP and used for transmitting/receiving the User Bit from the AES/EBU frame.

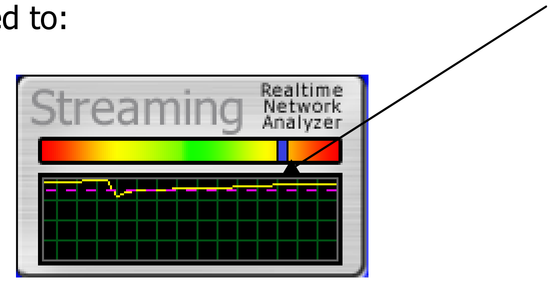
Regarding compatibility, Prodys are committed to work alongside the EBU NACIP group in the definition of the new standard for audio over IP. Keeping ahead of developments and supporting all protocols stated as 'mandatory' so far in the new Tech 3326 standard for audio over IP.

V.6.6 Proprietary (set of protocols) v2

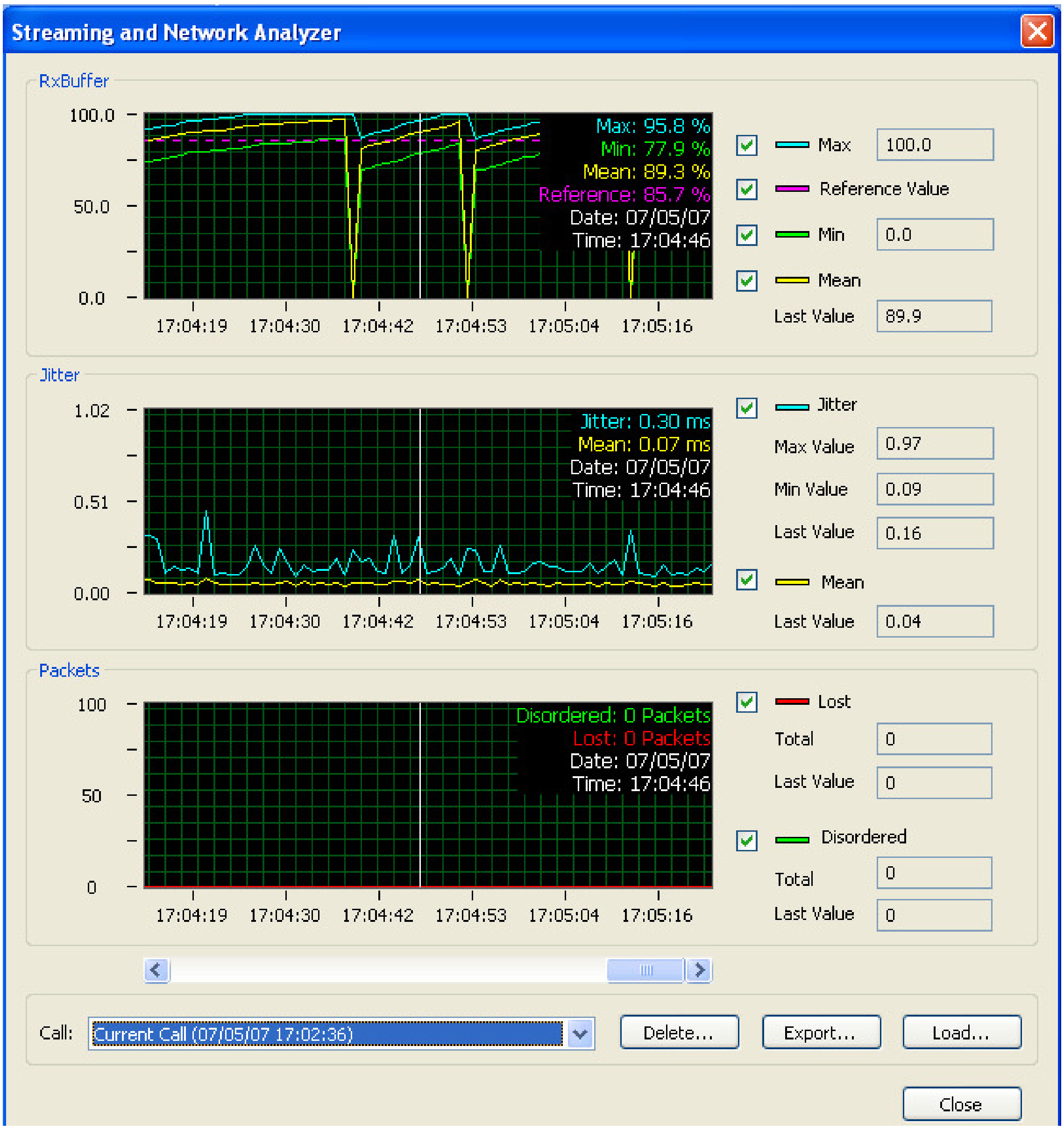
This set of protocols is based on the previous one, and it uses the same ports and different protocols. But, with this new protocol, it is possible to obtain network related parameters such as jitter, lost or disordered packets in real time during the audio connection. To be able to get all this information, Proprietary Protocol V2 should be configured as audio protocol. This protocol is not compatible with the previous one (version 1). Proprietary protocol version 1 will allow the user to get information about jitter and buffer usage, but will not allow the user to obtain information about lost and disordered packets.

This information will be saved separately for each connection in RAM memory. Up to 24 hours of data can be stored.

Once the connection is established, the user can access the 'real time network analyzer' by clicking on the 'buffer occupation graph', in order to get information related to:

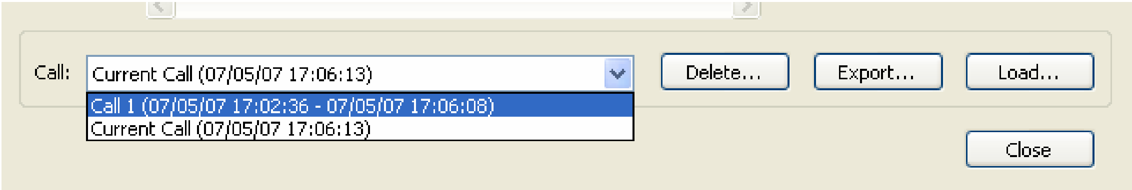


-Rx occupation: With average, maximum and minimum usage. Very low percentage of buffer occupation will cause audio interruptions and drop-outs.
-Jitter: High jitter values will match with low buffer occupation.
-Lost and disordered packets.



All the information is displayed in different graphs, synchronized with each other, so that the user can move through all the data very easily.

Data from each connection is stored independently so that it is possible to access data from connections other than the current one. In addition, it is possible to delete, export or import data from any previous call.



V.6.7 PRODYS PORTS for Prodys Proprietaty protocols (v1 & v2)

These are the **default** ports used by Prodys IP Codecs for their IP connections:

HeraFlash &
Prodys Control
UDP:50013

Web Page
TCP 80: HTTP
TCP 50011: Web Page
TCP 50017: ProdysControl

Audio Streaming
TCP 50019: P-RTCP (Control)
UDP 50021: P-RTP Unicast L1
UDP 50023: P-RTP Unicast L2
UDP 50025: PX-RTP Multicast L1
UDP 50027: PX-RTP Multicast L2
UDP 50037: P-AUXP Datos auxiliares (from version 4.8.0 on)
UDP 50039: P-UbP U-BITs (from version 4.8.0 on)

Test Streaming Tool
TCP 50033
UDP 50033

External Protocol (P-XP)
TCP 50031: Control Port
TCP 50035: Status Port

It is possible to change these ports from the web configuration window. Please, for more information about port changing, read chapter IV.5.1.3 – Ports.

V.7 SIP

The European Broadcast Union (EBU) is promoting the interoperability of audio codecs for any manufacturer. For this purpose the use and the application of a subset of the Internet Protocols has been proposed. This effort will allow to setup in a friendly way audio streaming communications between several vendors of equipment.

The deployment of SIP Protocol Servers among the network will support calling remote parties just by invoking their network name regardless of the actual public o private IP addresses. Neither is required to manage the office routers and the firewall's TCP/IP ports for any new communication path.

The physical location, often related to fixed IP addresses or subnets, is not further meaningful. In this sense, specially portable audio codecs will profit with an easy call procedure. Neither is required to agree ahead on the audio compression type and data stream rate, because the SIP Protocol manages by itself to negotiate the convenient communication details with the remote party.

For Prodys' former customers performing a call using SIP is much the same as using Prodys' proprietary protocols given that the SIP configuration is set.

SIP supports up to date only unicast streaming. Any PortaNet's audio compression mode is supported also if SIP protocol is in use.[28] For further details please refer to the "Technical Description" at this document.

- **SIP Port**: TCP/IP port dedicated to SIP signalling for establishing, updating or finishing a call. Port 5060 is mandatory by the standard if direct calls without gateways are expected.

- **Audio Port**: TCP/IP port dedicated to the RTP communication, this is the actual port for the audio streaming payload. By default the number 5004 is reserved.

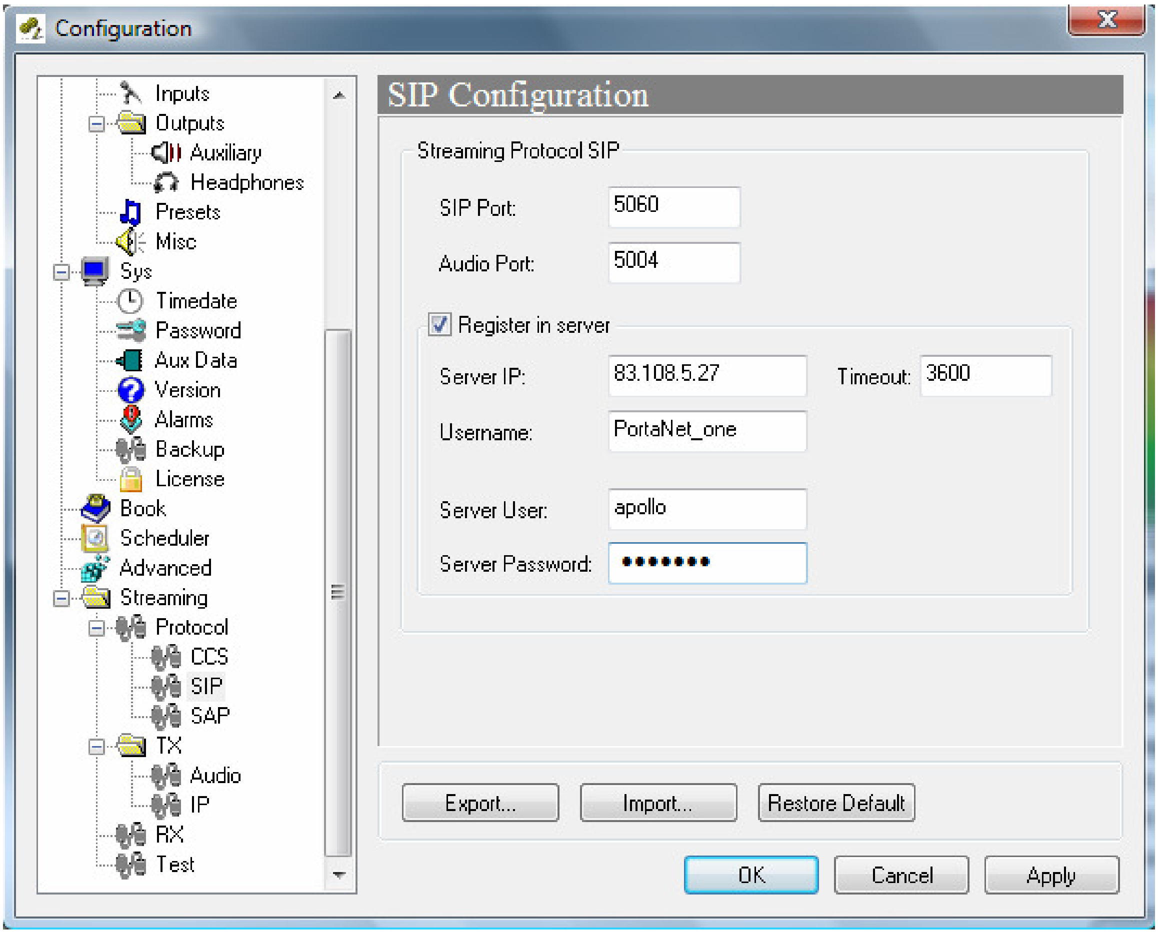
If SIP server support is granted by your network:
- **Server IP**: IP address of the SIP Server. At this point is registered periodically the proper user information for full protocol support.

- **Timeout**: Time in seconds before the user information registered at the previous server is flushed. Therefore the information about the SIP user are updated from time to time.

[28] The apt-X compression is still undefined by the EBU Tech 3326 standard. For compability reason this mode is not supported if SIP protocol is selected.

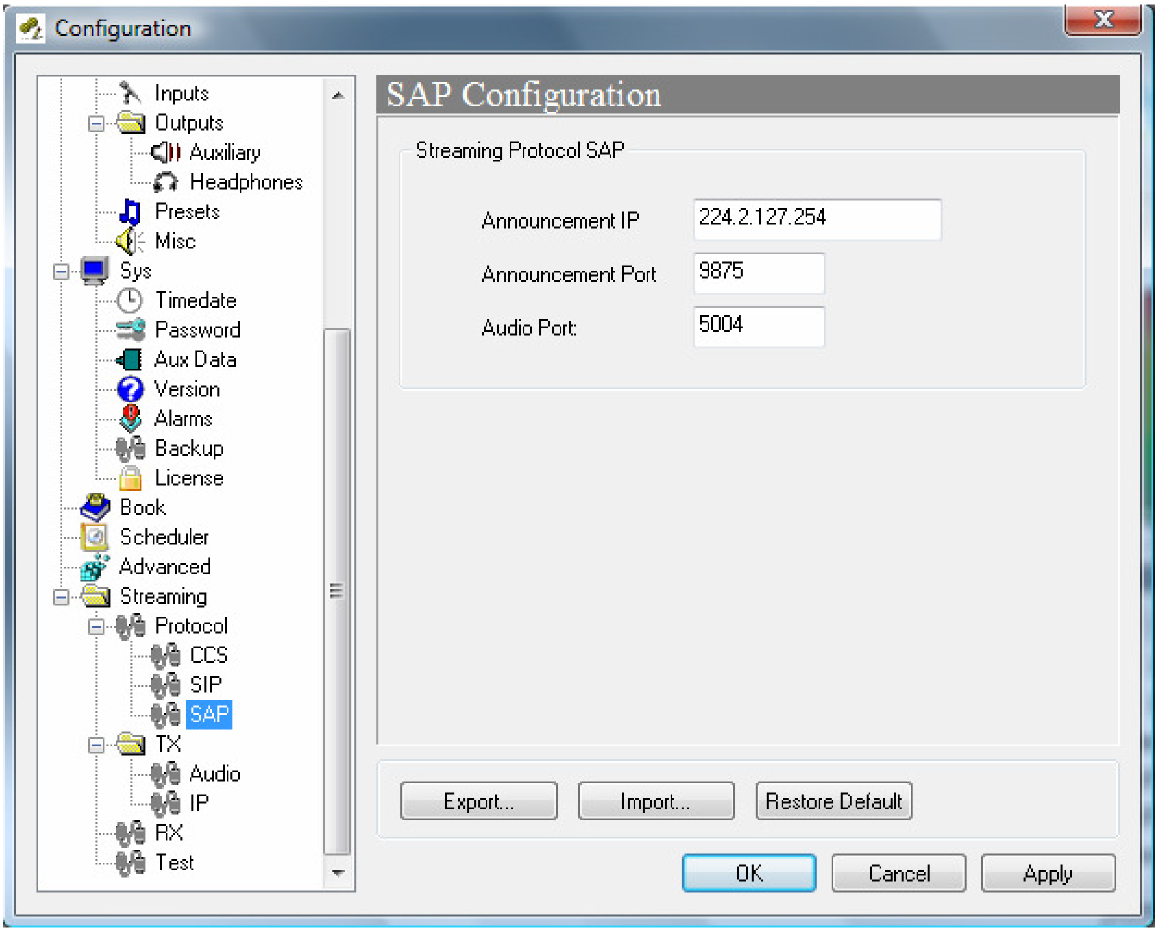
- **Username**: Your alias on the Internet regardless of your current IP address. By this alias you are identified for other SIP participants.

- **Server user & password**: Some SIP servers requires authentication before proceeds the register of SIP users.



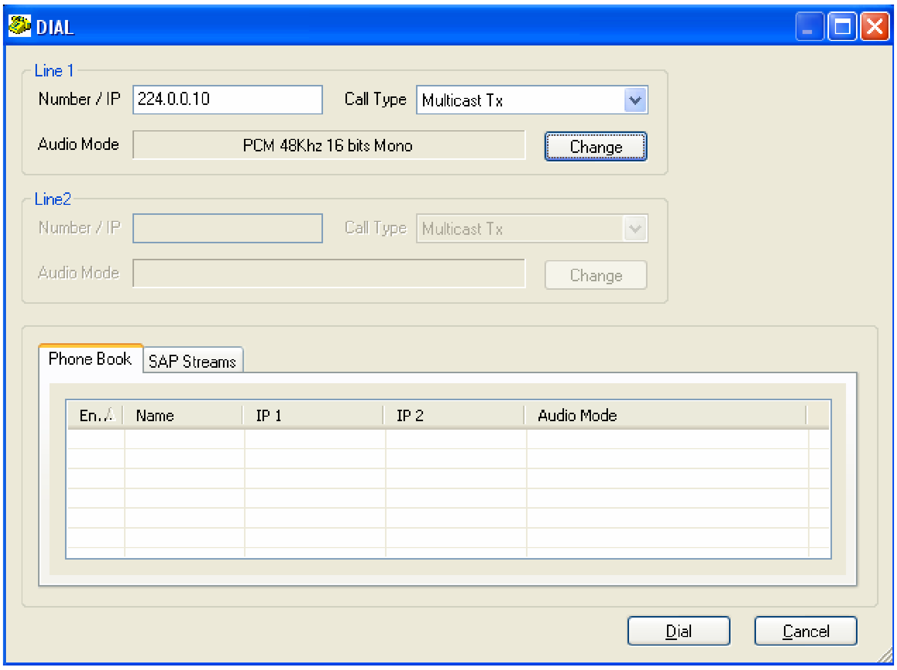
V.8 SAP

When selecting SAP as communication protocol, only multicast calls will be available (point-to-multipoint). In addition, the user will have to indicate the announcement IP address and port on which the audio streaming is 'announced'.

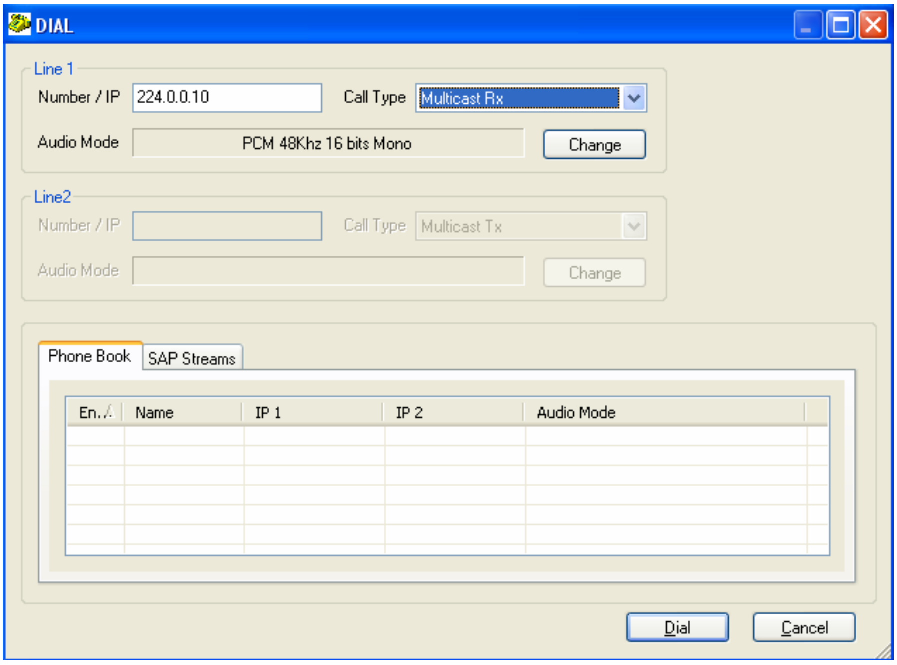


The default values for these parameters are the default ones defined by the SAP standard (RFC 2974). This address and port will be used by the transmitter to ‘advert’ its audio streaming broadcasting. In the case that these parameters are modified by the user, they should be configured to the same values at both ends, the sender and the receiver.

When making a call with SAP as communication protocol, only Multicast Tx and Multicast Rx call types will be available. To launch an audio streaming over IP with SAP the user shall select Multicast Tx as call type from the dialing window, and type in the multicast IP address to send the audio to.



To receive this audio streaming, the user shall select the Multicast Rx call type, and will introduce the same Multicast IP address as the transmitter.



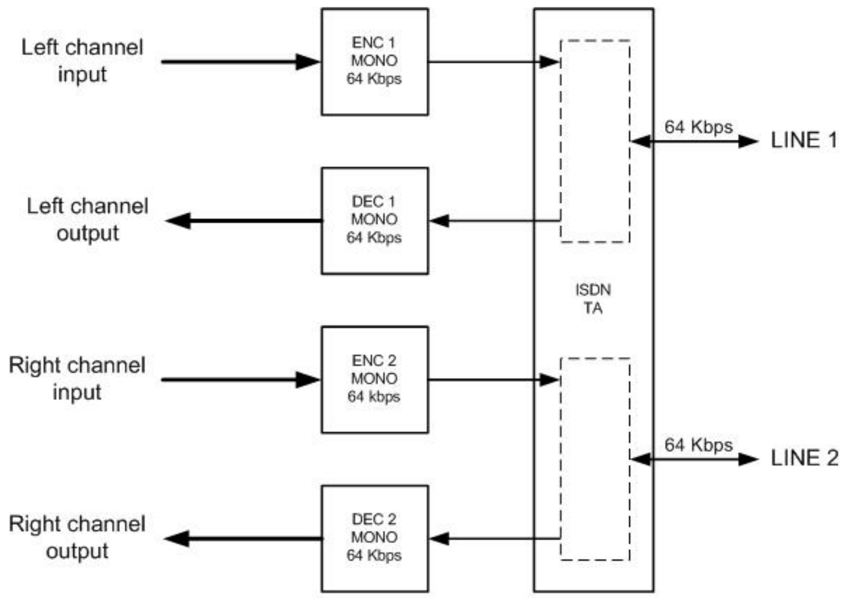
As many receivers as required can join this multicast address and receive the audio, without any increment in the required bandwidth, given that this communication is based on IP multicast technology.

V.9 Nereus operating as an ISDN codec

The Nereus as ISDN codec is very similar to the Pronto 3, therefore those users that are familiar with the Pronto 3 will find it easy to operate the Nereus over ISDN.
The most important points that a user must have in mind for this mode are as follows:

1. **ISDN interface:** The ISDN interface of the Nereus allows connection to a basic rate ISDN line. This interface allows the user two bi-directional channels with a bandwidth of 64 Kbps in each direction. There is an additional channel of 16 Kbps that is used for signalling. This is why a basic rate connection is sometimes represented as 2B+D. The Nereus identifies each B channel as Line 1 and Line 2 respectively.

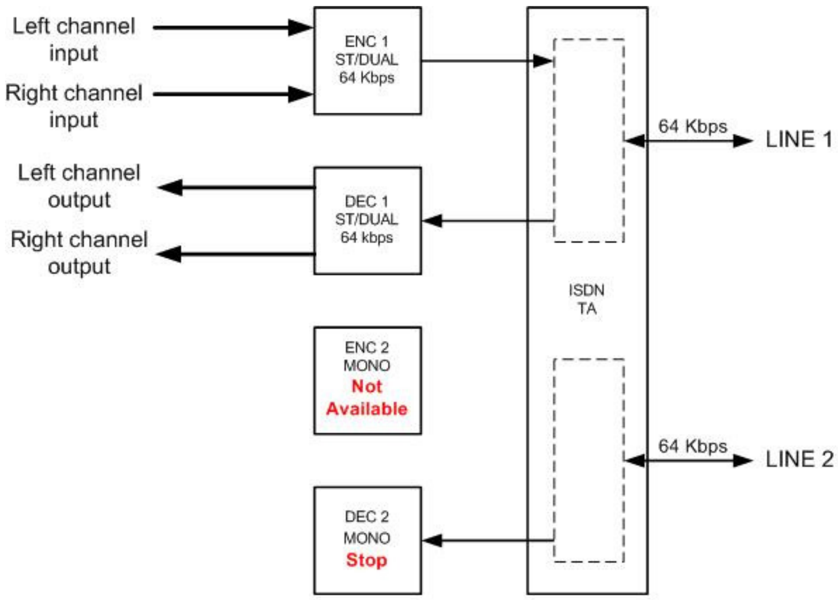
2. **Dual Codec:** The Nereus can operate as "DUAL CODEC", which means it can establish two totally independent communications via each B channel at 64 Kbps. This is only possible when each B channel or communication line uses a MONO mode compression algorithm in this situation the Nereus will be able to use two Encoders and two Decoders independently. This limitation is imposed by the fact that there is only one stereo audio input and one stereo audio output.



PRONTONET AS CODEC DUAL

The left audio I/O channel is always assigned to Line 1 and the right audio I/O channel is always assigned to Line 2.

3. **Communications in Dual mode, Joint Stereo or Stereo at 64 Kbps**: When we configure Encoder 1 to Dual mode, Joint Stereo or Stereo at 64 Kbps, Encoder 2 is disabled since there are no free audio inputs or outputs to assign to it.

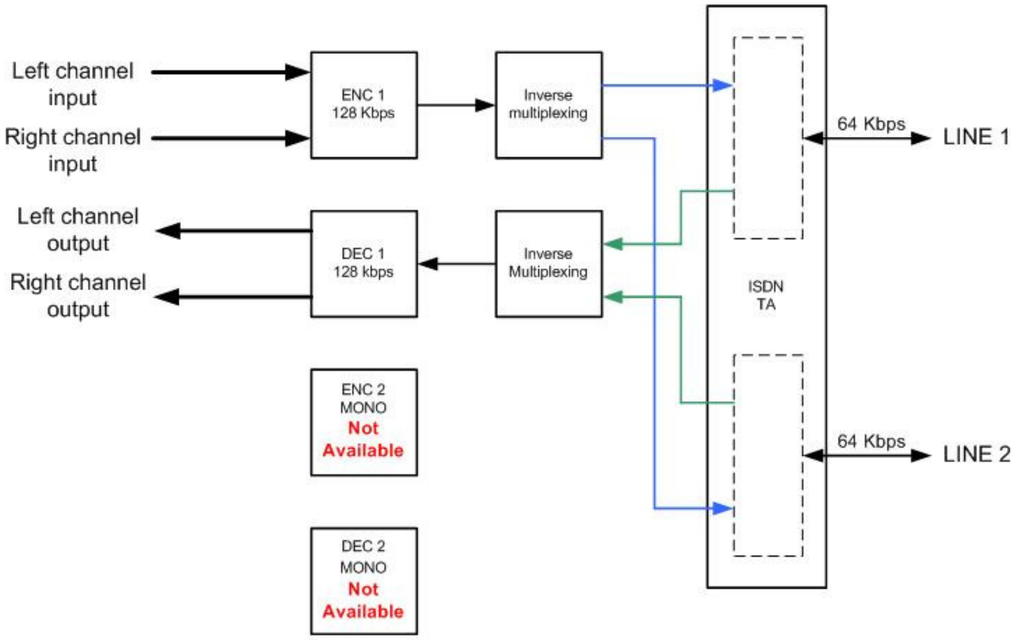


PRONTONET WORKING AT 64 Kbps IN STEREO/DUAL MODE

4. **Communications at 128 Kbps – Inverse Multiplex:**

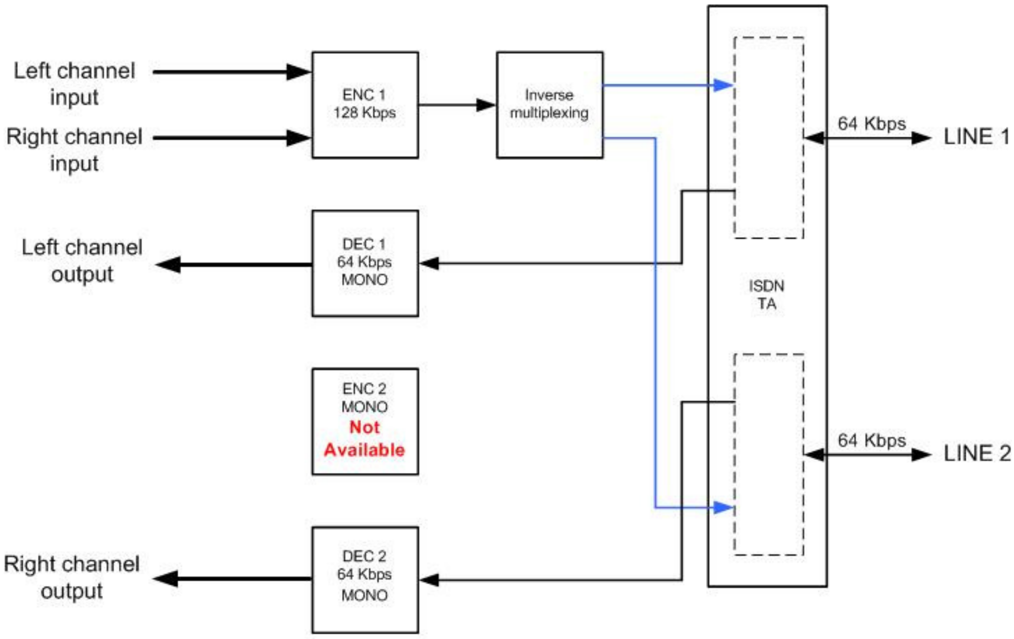
Certain algorithms permit the compression of the signal at bit-rates other than 64Kbps, for example at 128Kbps. With the Nereus it is possible to send or receive audio compressed at 128 Kbps using both the B channels of the basic rate ISDN connection. This is done by using 'inverse multiplex' techniques that split the transmission of the full 128 Kbps over two channels of 64 Kbps and then sum the parts together again at the other end.

Inverse Multiplex is often known as 'bonding'

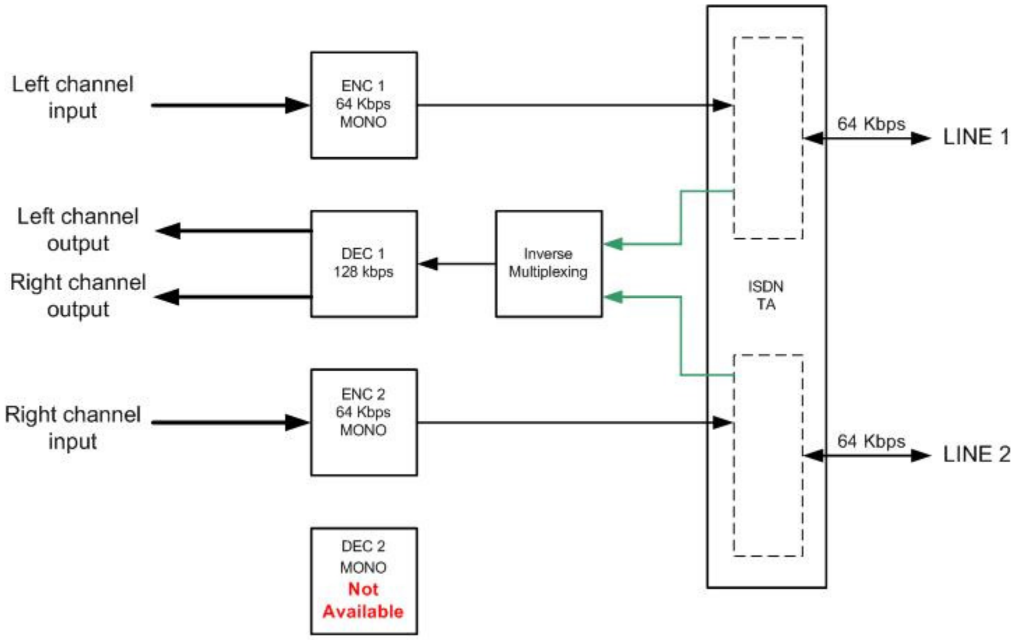


PRONTONET WORKING AT 128 Kbps

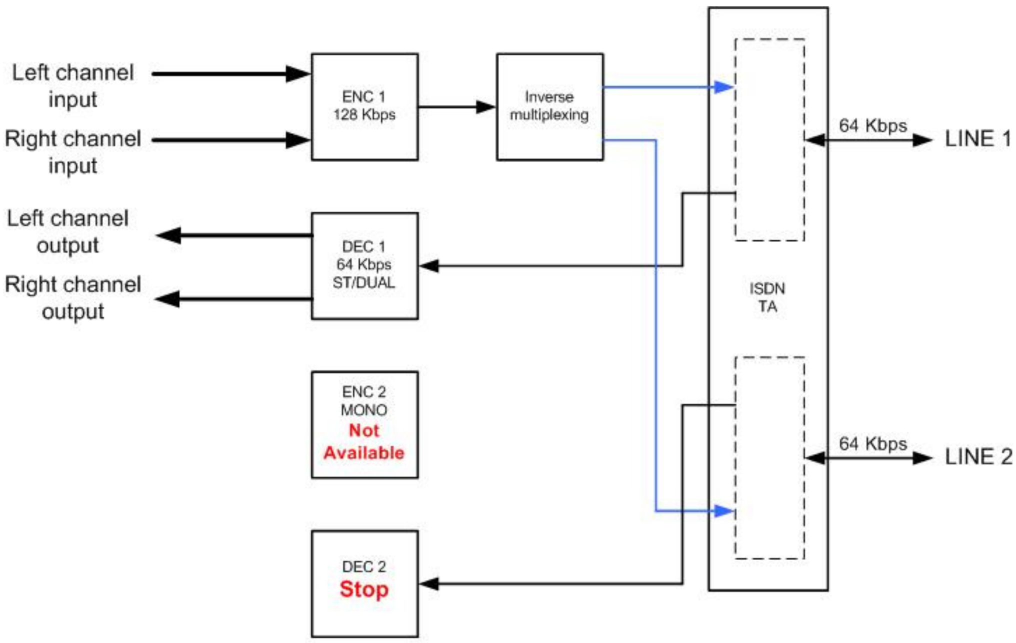
5. **Communications combining 128 and 64 Kbps**: It is possible to transmit encoding audio at 128 Kbps and to receive decoding audio at 64 Kbps or vice versa. Here below there are different examples of it.



PRONTONET WITH ENCODER AT 128 Kbps & DECODER MONO AT 64 Kbps



PRONTONET WITH ENCODER AT 64 Kbps & DECODER AT 128 Kbps



PRONTONET WITH ENCODER AT 128 Kbps & DECODER ST/DUAL AT 64 Kbps

V.9.1 Establishing ISDN calls

To make a call the CALL 1 and CALL 2 keys are used, or an entry from the Address Book is selected using the BOOK key. Bear in mind that the use of an entry in the Address Book will also reconfigure the Encoder and so is not limited to only making calls. We need to distinguish between calls at 64 Kbps and calls at 128 Kbps:

- Making calls at 64 Kbps

The CALL 1 key is for calling on Line 1 and the CALL 2 key is for calling on Line 2. On pressing the CALL 1 key the display prompts you to enter a number. Once the number is entered and the ENTER key is pressed, Nereus will go ahead and make the call.

The CALL 2 key operates in exactly the same way.

- Making calls at 128 Kbps

Each line needs to be connected independently using the CALL 1 and CALL 2 keys.

To disconnect a call press and hold the appropriate CALL key for more than one second.

V.9.2 Receiving calls via ISDN

In general you must bear in mind the following when dealing with incoming ISDN calls:

- **Automatic or manual response:** Incoming calls can be answered automatically or manually depending how the menu option ANS is configured for the ISDN port. If the ANS mode is set to manual, the appropriate CALL key must be pressed to accept the call and connect to it.

- **Call filters:** It is possible to record up to three numbers for each line that work as call filters, meaning that the line will only connect to calls that come from these pre-programmed numbers. This option is found in the ISDN set up menus under CNUM (Calling Number).

- **Local number**: It is also possible to assign a single number to each line in a way that the line will only respond to calls to this local number. This can be used if you need to map an ISDN directory number to a specific audio port. This option is found in the ISDN set up menus under LNUM (Local Number).

- **Receiving calls over G711 (Voice mode)**: the ISDN indicates calls on G711 as voice mode. The Nereus recognises this type of incoming call

and proceeds automatically to reconfigure the Encoder and Decoder to this mode. As such, the user need not configure anything to allow the automatic reception of incoming calls using G711.

The Law is selected automatically, depending on the ISDN protocol selected – A Law for Euro ISDN and U-Law for all others.

V.9.3 Restrictions in ISDN communications

As we have already stated, Nereus can work as a DUAL CODEC over ISDN, meaning it can operate with two independent lines. However, this is only possible when two MONO communications are used. The use of Dual, Joint Stereo or Stereo modes creates certain restrictions on the system that are in place to avoid conflicting situations.

1. DUAL, JOINT STEREO or STEREO modes are only available on Encoder 1.

2. When Encoder 1 is configured to DUAL, JOINT STEREO or STEREO, Encoder 2 is not available.

3. **Nereus will connect the line but the audio will not be decoded in the following cases**:

    a. The line 2 is connected and the Decoder 2 is FRAMED. Nereus receives a call in the line 1 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

       Line 1 will connect but the audio will not be decoded since Nereus working as a DUAL CODEC only allows two MONO communications. The display shows an error code to indicate the situation.

    b. Nereus receives a call in the line 2 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

       Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows an error code to indicate the situation.

    c. The line 1 is connected and the Decoder 1 is FRAMED in DUAL, JOINT STEREO or STEREO mode at 64 Kbps. Nereus receives a call in the line 2.

       Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows "**STOP**" to indicate the situation.

```
L1→CONNECTED FRAMED
L2←CONNECTED STOP
```

V.10 Nereus operating as an X21 codec

This operation is very simple. Remember that the Nereus only has one X21 port and that it is possible to connect or disconnect simply using the CALL 1 key. As soon as the connection is made the Nereus starts to transmit and receive audio via the X21 port.

V.11 How the backup mode works

Nereus when is configured as IP codec or X21 codec (NET = IP o NET = X21) can use the ISDN connection as backup line, that is, the ISDN will be an alternative communication line when the IP or X21 line is dropped.
This option is available in the SYS menu:

```
     SYS MENU      »
{LOOP}PLL BACKUP FAN
```

V.11.1 MASTER & SLAVE Configuration

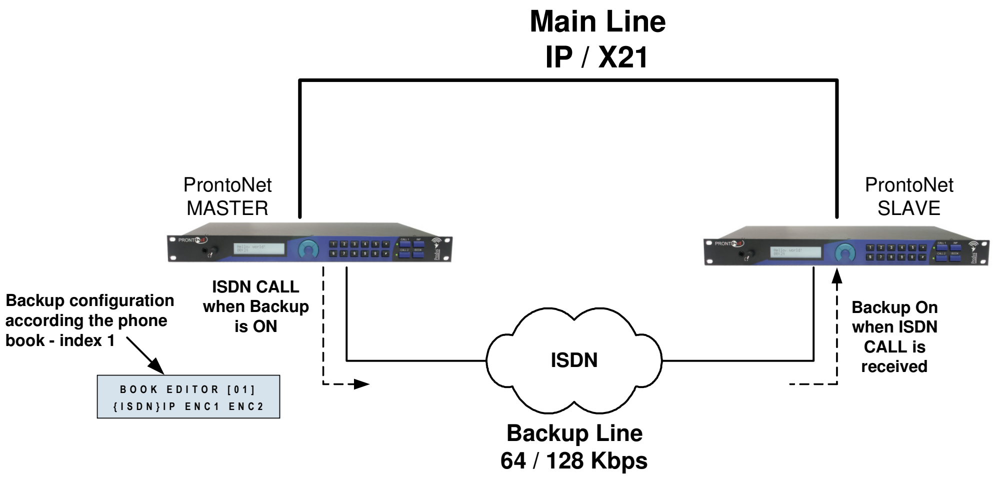
The backup operation is based on the interaction between two Nereuss, one of them working in MASTER mode and the other end configured as SLAVE. The user must configure each unit in one of these modes before to enable the backup facility. The option SYS – BACKUP - MODE allows the selection of Master or Slave.

Once the backup option has been validated, the Nereus configured as MASTER monitors continuously if the main line is working appropriately in order to decide if the main line is dropped or, in case the unit is working in backup mode (ISDN connection), if the main line (IP or X21 communication) has been re-established.

The Nereus MASTER uses the information stored in the index 1 of the phone book to configure the unit and to call to the stored numbers.

The Nereus SLAVE only works in backup mode depending on if there is communication by the ISDN line.

Please, find below an example of Nereus backup application to note the differences between both working modes.



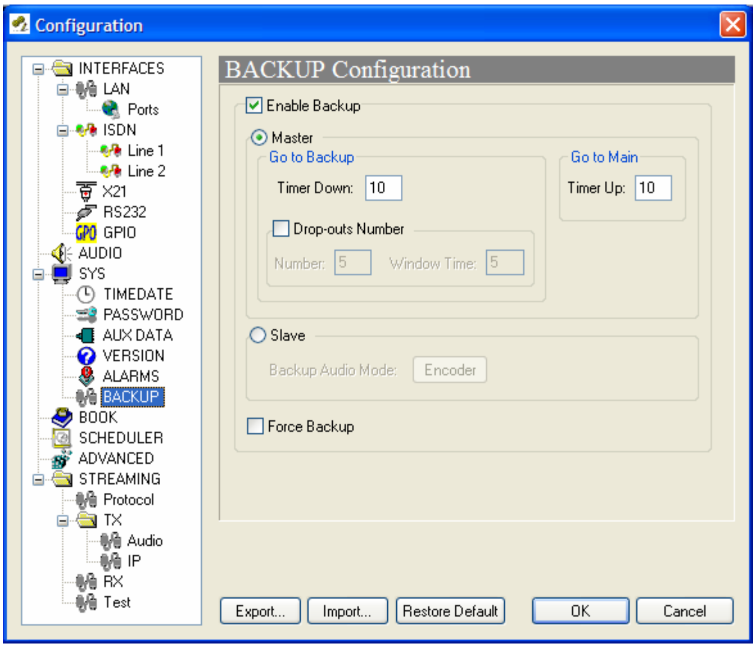
As we can see, the Nereus configured as MASTER monitors the main line and it decides when the backup ISDN line must start to work. The Nereus SLAVE only works on ISDN mode depending on if there is communication by it.

The user must be careful to disable the backup mode (BACKUP_ENABLE_OFF) to avoid that there was any interference in the installation and configuration operations of the unit.

V.11.1.1. Nereus MASTER operation

As we mentioned before, the Nereus configured as MASTER monitors continuously if the main line is working appropriately once the backup option has been validated in the menu (BACKUP-ENABLE-ON). The Nereus MASTER decides if the main line is dropped when the audio synchronism has been lost for the programmed time in TIMER-DOWN option of the menu or when as many drop-outs as defined in the Drop-outs option of the menu have occurred within the period of time configured by the user[29].

[29] This criteria for the BackUp operation was included on version 4.8.1.



When the audio synchronism has been lost during the programmed time by the user (TIMER-DOWN), or there have been as many drop-outs as defined by the user within the "window time", the unit will be configured in backup mode according the encoding configuration stored in the index 1 of the phone book. Once it has been configured, the codec will proceed to call to the programmed numbers in the same index the number of calls (one or two B channels) will depend on the mode that the user has chosen (64 o 128 Kbps) when the unit starts working in backup mode.

Nereus will try the connection as many times as it will be necessary until the connection will be established or until it decides that the main line has been recovered. If the backup is disabled during the calling process or when the ISDN communication is connected, the unit will proceed to work at IP or X21 mode independently if the main line is recovered or not.

While the unit is working in backup mode, the unit is monitoring the main line in order to decide if the line has been recovered or not. If the audio synchronism is detected in the received audio by the main line, the programmed countdown in the TIMER-UP starts. If the countdown finishes, Nereus goes to IP or X21 mode again and therefore leaving the backup mode after ending the communication by the ISDN line.

In order to avoid the entry of unwanted calls, Nereus terminal adapter allows the programming of call filters (BACKUP-TA_CNUM).

V.11.1.2. Nereus SLAVE operation

If Nereus works as SLAVE, the entry or exit to the backup mode will be guided by different criterions than the defined ones in the MASTER mode. First of all, it is necessary that the Backup option be enabled.

The unit will work by default in IP or X21 mode monitoring the state of the ISDN line continuously. If an incoming call is detected, the unit will switch to backup mode automatically. If the call hungs up, the unit will pass to work in IP or X21 mode again.

During the backup phase the Nereus SLAVE goes on sending coded audio through the main line in order for the Nereus MASTER can detect that the line has been recovered.

It is also possible to program call filters in the Nereus SLAVE in order to avoid the unit answering unwanted calls.

Nereus MASTER Backup Configuration

1.- Disable the backup operation: SYS - BACKUP – ENABLE - OFF

2.- Set Nereus as MASTER: SYS – MODE - MASTER

3.- Set the timers: SYS – TIMER – DOWN – x sec.

SYS – TIMER – UP – y sec.

4.- Set the backup audio mode and the ISDN numbers:

CONF – BOOK – INDEX 1 →

```
  BOOK EDITOR [01]
{ISDN}IPENC1 ENC2
```

5.- Enable the Backp operation: BACKUP – ENABLE - ON

Nereus SLAVE Backup Configuration

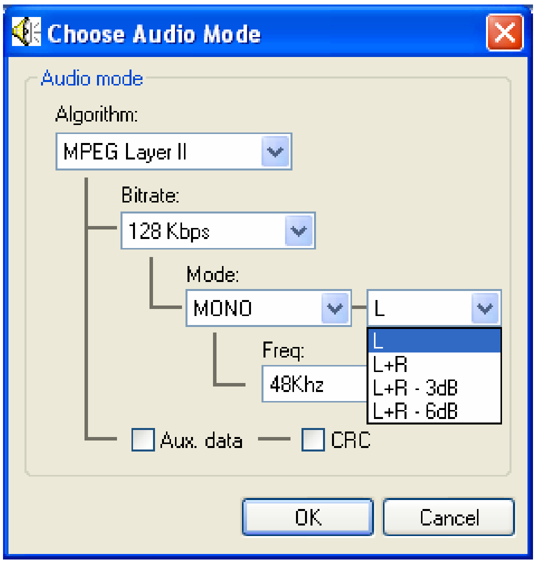
1.- Disable the backup operation: SYS - BACKUP – ENABLE - OFF

2.- Set Nereus as MASTER: SYS – MODE – SLAVE

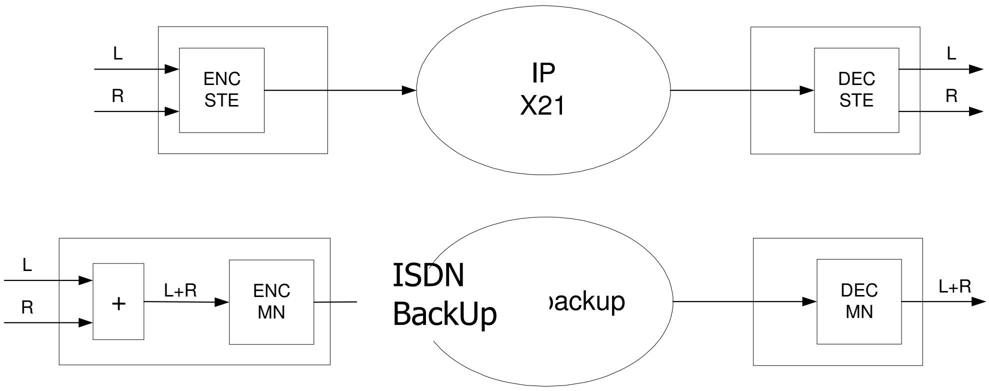
3.- Enable the Backp operation: BACKUP – ENABLE – ON

It is possible to select between the left channel L or L+R as audio source of the encoder 1, and R or L+R as audio source for encoder 2, when mono mode is selected for the backup (ISDN) connection.

This option is only available from the Nereus web Page:



This option is very useful for an 64kbps ISDN link, where sending a stereo signal could involve a lack in the audio quality. Thus, encoding L+R signal, as a mono signal, will provide a much better audio quality. The diagram below explain this:



Problem-solving guide

This chapter is aimed at providing some solutions to common problems when using Prodys Codecs.

VI.1 Audio problems

Audio problems and some possible solutions:

VI.1.1 No Audio on the outputs

When audio is not present on any output and before proceeding, it is convenient to check the audio matrix configuration in order to make sure that the audio sources has been correctly selected.

VI.1.2 The program line is connected but there is no audio on the outputs

In order that audio outputs carry audio, the corresponding decoder must be synchronized, that means, the decoder has to detect the format of the bit stream from the audio data being received. This auto-detection is automatic, and this situation is represented in the screen with the word "FRAMED".

```
L1→CONNECTED  FRAMED
L2:IDLE
```

Thus, when the decoder is not 'framed', there is no audio on the outputs.

Some things that can cause the decoder not to be synchronized:

1. The communication line is not working properly.
2. The encoder at the other end is not configured in a compatible mode, or it is not encoding properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything works correctly.

If the decoder is 'framed' but there is still no audio on the outputs:

1. There is no audio present on the inputs of the codec at the other end, or the audio in the other codec is not well configured.
2. The codec at the other end is not working properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything work fine.

VI.1.3 There is no audio output at either end

Some things that can cause these problems are the following:

1. The communication line is failing.
2. The remote codec is not working properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything work fine.
4. The codecs involved in the communication are not compatible.
5. Both codecs are configured in AUTO mode (automatic encoder mode).
6. The audio settings are wrong configured in both codecs.

VI.1.4 Audio distortion

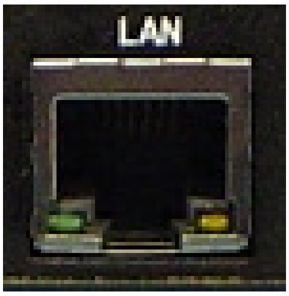
It is possible to modify the gains on inputs and outputs, so check if this has been set properly. The audio input and output levels can be monitored from the display by pressing the INF key, or from the web page.

Another cause could be that the other end is using digital audio such us AES/EBU as an audio source, and no AES/EBU signal is being sent, or the equipment is faulty.

VI.2 IP communication problems

VI.2.1 Prodys Codec's Web Page cannot be accessed

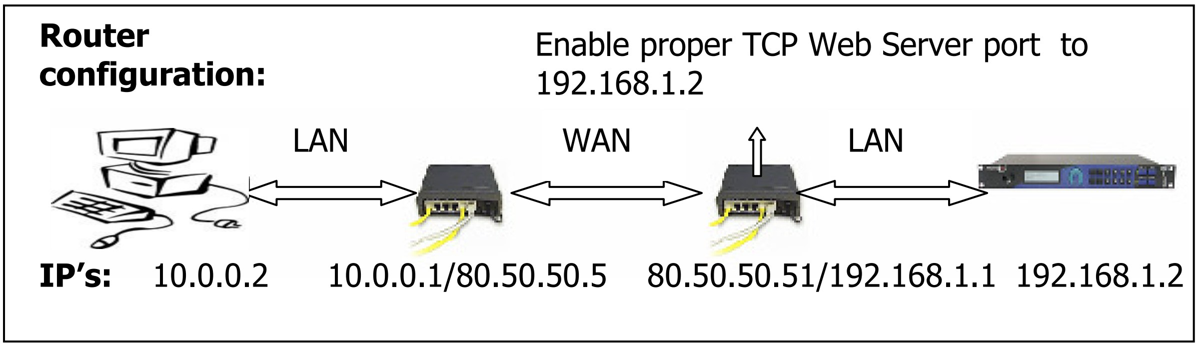
1. The cable or the connector are faulty. There are some leds on the rear panel to check the Ethernet link:



- Green LED → LINK STATUS: ON = Connected
- Orange LED → RECEIVE STATUS: On = Receiving Data.

2. The IP address of Prodys Codec and the PC from which we are trying to access the web page are not in the same subnet, even although they are connected to the same LAN. The default factory settings for the IP address and netmask are 192.168.100.100 and 255.255.255.0 respectively. The user must change the IP settings in the computer or in Prodys Codec to match the same network.
   To change the IP address of Prodys Codec, the user can use the control keypad or the web page. In the menu, it is set by selecting CONF-PORTS-LAN. The IP settings can be entered manually or they can be obtained automatically when the unit starts from a DHCP server.
   Once the IP setting on the PC and the Prodys Codec are configured properly, you can check that IP connectivity exists by typing the following command at the command prompt of the operating system: C:>ping 192.168.100.100 ↵. This tool will inform the user whether there is IP connectivity between the PC and Prodys Codec or not. If there is IP connectivity bur the problem still persists please refer to point 4.

3. The PC and Prodys Codec are not connected to the same LAN, but connected through a router. This connection could be, for example, a connection over the Internet with DSL routers. In this case, to access and operate the Prodys Codec, the router should open the a “Web Server Port” (this is by default HTTP port 80) and a range of 30 TCP ports from the “Base Port” on (default from 50011 until 50041) and forward this traffic to the Prodys Codec IP address. See pop up window of chapter **¡Error! No se encuentra el origen de la referencia.**-PORTS for setting these equipment parameters.



4. Microsoft Internet Explorer blocks ocx installation when trying to access Prodys Codec web page from microsoft internet explorer. The first time the user accesses the Prodys Codec web page, an OCX file has to be downloaded and installed in the computer. This is done automatically unless the web browser disables it. So, depending on the configuration of

the web browser, the following message can appear when first accessing the Prodys Codec web page:

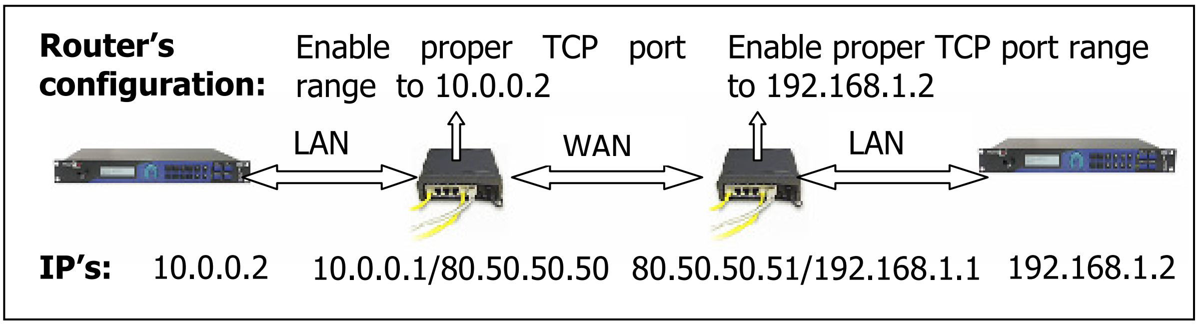
Go to Internet Options in IExplorer, click on 'Security' tab, and set 'prompt' when downloading ActiveX signed and unsigned controls at Local and Internet zones.

Windows Vista: Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, please disable UAC (User Access Control) on Windows Vista. Once the OCX file has been installed in the computer, UAC can be enabled again.

Each firmware version might have a different OCX file, so the new OCX should be installed as it is done for the first access to the web page of the unit. If the unit was upgraded and, depending on the 'cache' configuration of the Internet explorer, there might be problems when accessing the web page, given that the old web page might be offered by the browser instead of the real one, which should be installed to replace the old one. In this case, a message indicating 'Incorrect Versions' will appear as soon as the user click on 'Control' or 'Monitor' on the login page. Click on F5 to skip the cache entries, and access to the 'real' web page. Even after pressing F5 and, depending on the IExplrorer configuration and/or version, this situation might continue. In that case, go to Internet Options in IExplorer, click on 'General' tab, and delete temporary files.

VI.2.2 When connecting two audiocodecs in unicast, there is no audio at one end.

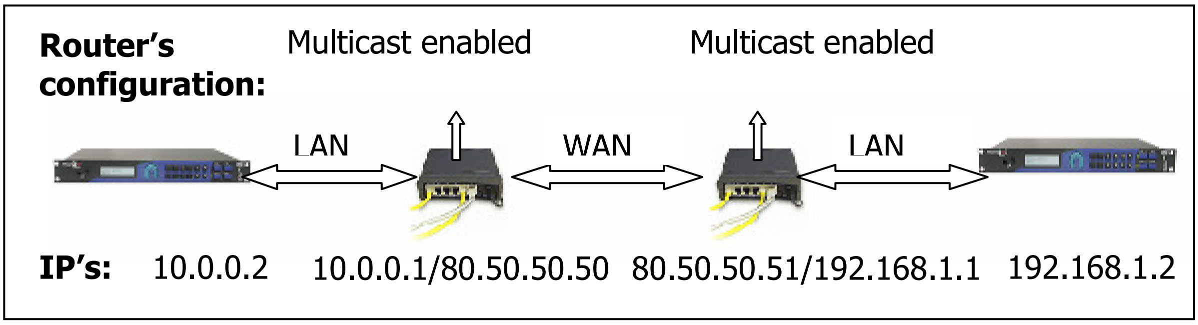
Please check the enabling of the full TCP port range ("Base port" + following 30 ports) of any router or firewall between the audiocodecs. Hint: start checking from the disturbed audiocodec on.



VI.2.3 No audio when connecting two audiocodecs using Multicast

1. Please check the enabling of the full TCP port range ("Base port" + following 30 ports) of any router or firewall between the audiocodecs.

   Take into account that only a small part of the Internet called Mbone supports multicast traffic, so to send multicast traffic over the Internet, a technique called 'tunneling' must be used. VPN networks can be used for this purpose.

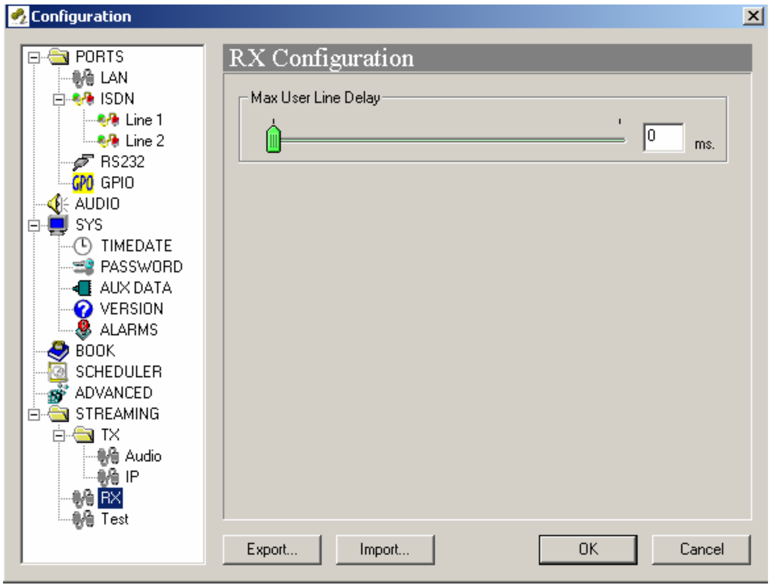


VI.2.4 Interruptions to audio when connecting two Prodys Codecs.

1. A decisive factor in real time audio streaming is the 'jitter', or delay variation. To deal with the jitter in the connection, PRODYS provides a tool which allows the user to modify the size of the reception buffer, and so, to compensate for the jitter. The maximum value for this buffer is 10 sec. This

buffer has to be configured from the web page to, at least, the same value as the 'jitter' in milliseconds.

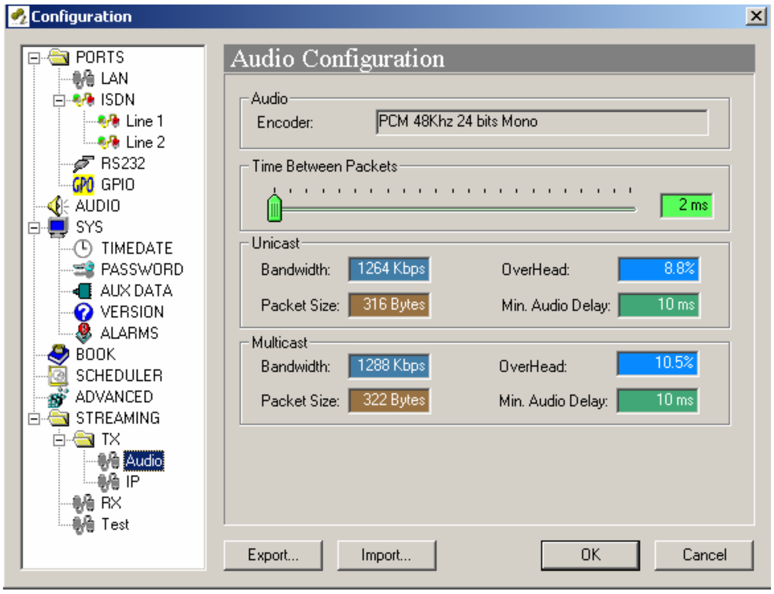
The 'jitter' can be measured from the 'Test streaming' tool provided from the web page.



Note: With the Real Time Network Analyzer, the user can get information about the jitter during the audio connection in real time. For more information about this, please refer to chapterV.6.6– Proprietary protocols v2.

2. The audio interruptions could be due to a reduction on the bandwidth in the IP connection. With the 'Test Streaming' tool from the web page, the user can measure the download and upload bandwidth between two Prodys IP Codecs. Once the user knows the available bandwidth, it is possible to select the proper bit rate for the compression mode.

   The user can obtain information about the delay of the encoder/decoder process for any particular mode, and the actual bandwidth which will be required for that mode, from the web page, in the 'audio' tab. Besides, in the non-block modes, like PCM or apt-X, it is possible to modify the size of the frames from 2 to 24 msc. In the rest of the modes, this size is fixed and determined by the corresponding standard. The larger the block size, the higher the delay, but the smaller the required bandwidth (more efficient use of IP packets), and vice versa.



Lost and/or disordered packets might be the cause of audio interruptions. To obtain information about these parameters in real time, the Real Time Network Analyzers can be used. For more information about this, please refer to chapter V.6.6 - Proprietary protocols v2

Appendix A

Technical Specifications

VII.1 Audio Interfaces

Stereo Audio Inputs:

- Balanced Analog inputs:
  Maximum input level: +22 dBu.
  Input Impedance: 20 Kohm.

- Digital inputs:
  AES/EBU format: EIAJ CP-340 tipo I/IEC-958 Pro
  Rate Converter: 1:3 to3:1.

Stereo Audio Outputs:

- Balanced Analog Outputs:
  Maximum output level: +22 dBu.
  Output Impedance: 50 ohm.

- Digital Outputs:
  AES/EBU format: EIAJ CP-340 tipo I/IEC-958 Pro
  Rate Converter: 1:3 to 3:1.

Audio properties*:

THD+N<0.0035%
S/N > 94 dB typical.
Crosstalk > 94 dB.
Phase Difference < 0.3º.
Quantification: 24 bits.

***With a tone of +22 dBu, Fs=48 Khz, 24 bits**

VII.2 Compression

- G722.
- G711 A/µ Law.
- MPEG 1,2 layer II (ISO/IEC 11172-3 /13818-3).
- MPEG 1,2 Layer III (ISO/IEC 11172-3 /13818-3).
- MPEG 2 AAC LC (ISO/IEC 13818-7).
- MPEG 4 AAC LC, LD & HE (ISO/IEC 14496-3).
- Standard and Enhanced aptX™.

VII.2.1 BANDWIDTH (KHz)

Legend:
x = Not available in the standard.
#,*,- = Not implemented

- **MPEG 1,2 LAYER II (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	4	X	X	4,9	x	x	7,3	#	#
64	10,7	4	4,8	11,7	4,9	6,1	11,3	7,3	11,3
128	20	10,7	16,3	15	11,7	13,6	11,3	11,3	11,3
192	20	14,5	20	15	15	15	x	x	x
256	x	20	20	x	15	15	x	x	x
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo /Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 1,2 LAYER III (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	8,1	#	#	8,2	#	#	8,1	#	#
64	15,2	8,1	8,1	15	8,2	8,2	11,3	8,1	8,1
128	18,2	18,2	18,2	15	15	15	11,3	11,3	11,3
192	20	20	20	15	15	15	x	x	x
256	20	20	20	15	15	15	x	x	X

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 2 AAC LC (ISO/IEC 13818-7)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	15,8	7,5	8,3	14	8	12	11,3	8,3	10,5
128	20	15,8	15,8	15	14	15	11,3	11,3	11,3
192	20	15,8	15,8	15	14	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

- **MPEG 4 AAC LC (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	18	7,5	8,3	15	8	12	11,3	8,3	10,5
128	20	18	20	15	15	15	11,3	11,3	11,3
192	20	18	20	15	15	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
64	7,5	7	7
128	x	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 4 AAC LD (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	10,6	#	#	11,8	#	#	11,3	#	#
128	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
192	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
256	20	18,4	18,4	15	15	15	11,3	11,3	11,3
384	20	18,4	18,4	15	15	15	-	-	-

- **MPEG 4 AAC HE (ISO/IEC 14496-3)**

Bit Rate	Fs=24KHz			Fs = 16 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
24	12	#*	#	12,3	#	#
32	16,8	#	#	15	#	#
48	16,8	12	15	15	12,3	15
56	16,8	12	15	15	12,3	15
64	20	16,8	20	15	15	15
128	20	20	20	x	15	15

- **aptX™ SD & ENH 16 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz		Fs=16 KHz	
	MN	ST/Dual	MN	ST/Dual	MN	ST/Dual
64	x	x	X	x	7.5	X
128	x	x	15	7.5	X	X
192	20	x	X	x	X	X
256	x	x	X	15	X	X
384	x	20	X	x	X	X

- **aptX ENH 20 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
160	x	x	15	x
320	x	x	X	15
240	20	x	X	X
480	x	20	X	X

- **aptX ENH 24 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
192	x	x	15	x
384	x	x	X	15
288	20	x	X	X
576	x	20	X	X

- **G722**

Bit Rate	Fs=16KHz
	MN
64	7.6

- **G711**

Bit Rate	Fs=8KHz
	MN
64	3.8

VII.3 IP Protocols and compatibility

- DNS
- HTTP
- ICMP
- IGMPv2
- IPv4 /TCP /UDP :RFC 791, RFC 793, RFC 768, RFC 1112
- RIPv2
- RTP
  - :RFC 3550, RFC 3551, RFC 2250, RFC 3119,
  - :RFC 3190, RFC 4184, RFC 3555, RFC 3640
- SAPv1 :RFC 2974
- SDP :RFC 4566, RFC 3264 (EBU Tech 3326 Standard)
- SIPv2 :RFC 3261
- SNMPv2
- SNTP

VII.4 Communications Ports

VII.4.1 ISDN[30]

- Protocols: EISDN, AT5ESS, DMS100 and NAT.
- 1 BRI connection. S/T and U interfaces.
- **BackUp system:** ISDN as a BackUp for IP or X21 links.
- RJ45 connector.

VII.4.2 X21 Port[31]

- Serial Synchronous interface.
- Bit rates: 64, 128, 192, 256, 384 and 576kbps.

VII.4.3 LAN port

- 10/100 Base-TX Ethernet.
- Connector type: RJ-45

VII.4.4 GPIO Port

- 4 TTL inputs and outputs.
- Inputs: Closure to ground.
- Outputs: Open collector. 40 mA Max o 40 VDC max.
- Connector Type: RJ45

VII.4.5 RS232 Port

- RS232 bi-directional asynchronous
- Supports data rates to 38.4 Kbps.
- Connector Type: RJ45

VII.5 Power Supply

- Universal power Supply
- Operating Voltage: 94-250 V
- Operating Line frequency: 47-65 Hz.
- Power Consumption < 150 W.

[30] Available on the optional ISDN/X21 module.

[31] Available on the optional ISDN/X21 module.

VII.6 Dimensions

- 3 U – 19” Rack Mount.
- Depth: 363 mm.

VII.7 Environment

- Temperature: 0 – 50ºC.
- Humidity: 10 to 90% non-condensing.

Appendix B

Updating the firmware

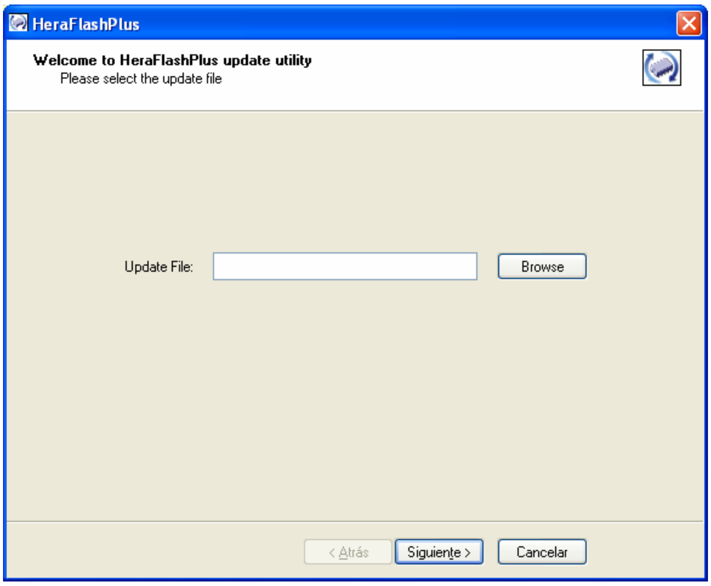
HeraFlashPlus application allows the user to update the firmware of several Nereus card simultaneously.

From version 5.2.1 and by using HeraFlashPlus version 2.5.0 or higher, the upgrading can be performed over WAN networks, such as Internet

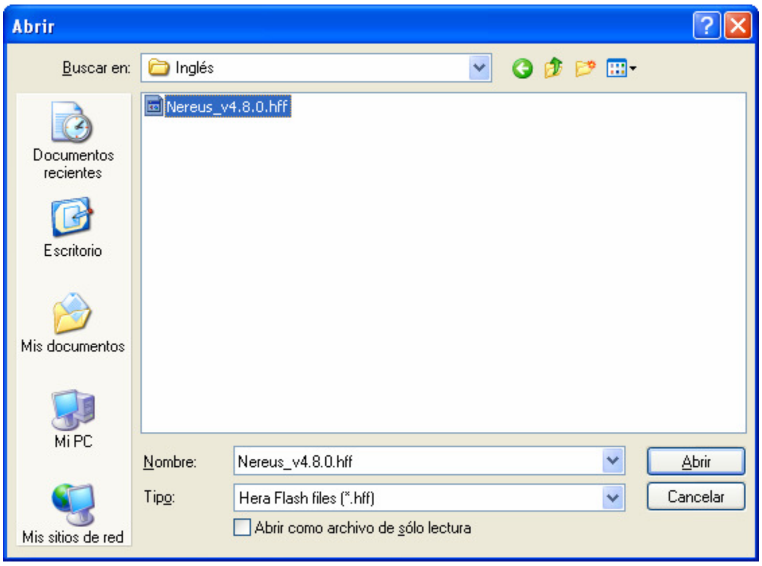
From version 5.4.1 onwards, all Prodys IP codecs will share the same firmware file, making it easier to upgrade several different units to the latest firmware version.

Follow these steps to upgrade them:

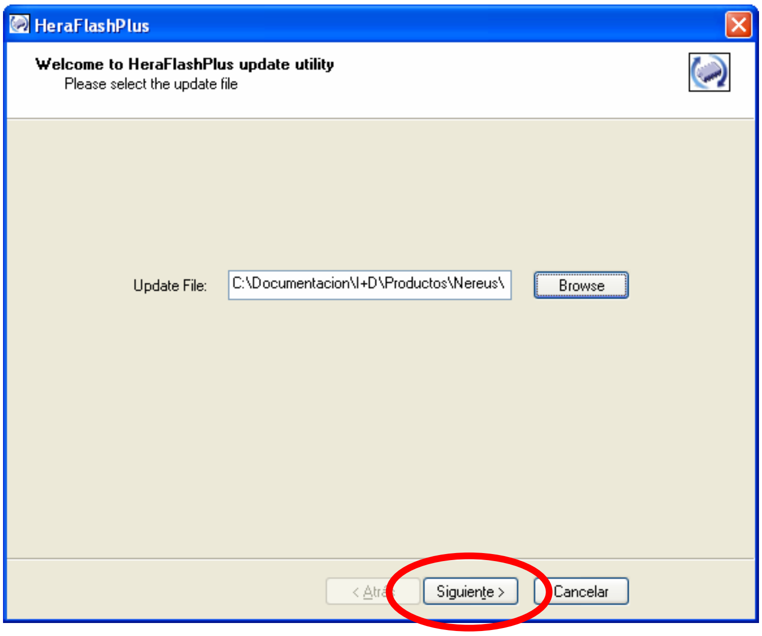
1. Start up the HeraFlashPlus application



2. Press the ***Browse*** button and select the update file. The selection dialog box will only show files with the extension .hff, representing the permitted update files.

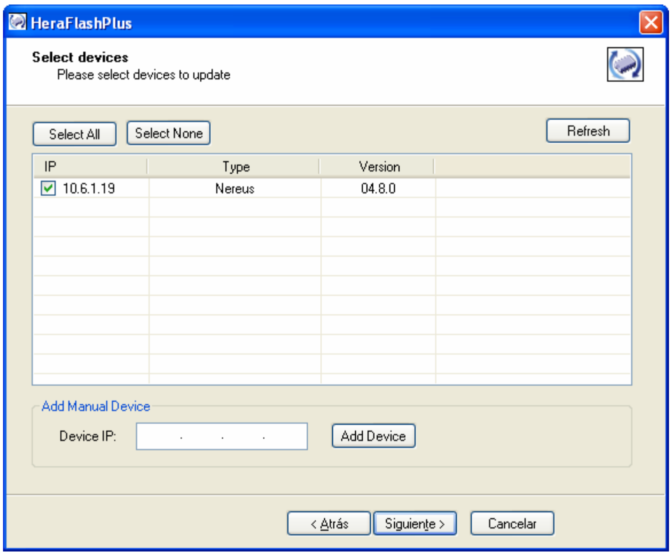


3. Once the file has been correctly selected, press Next to go to the devices selection dialog

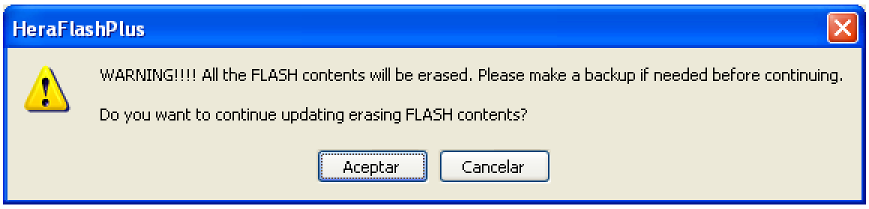


4. The selection devices window shows the devices connected to the network that have been automatically detected. The key refresh allows to repeat the detection process.

> **The network must support broadcast traffic. If the broadcast traffic is not allowed, the IP addresses must be introduced manually by writing the IP address of each card in the Device IP box and clicking the Add Device box.**



5. To select a device that will be updated, enable the check box that is close to its IP address.
6. Once all devices has been selected, you will see a warning message telling you that all the contents of the Flash will be erased and that you must back-up this data BEFORE updating the device.



7. If you go ahead, the update process will begin, the Flash will be erased and a progress bar will tell you more info and will alert you when it is finished.